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# ACCESS CONTROL IN FULL-DUPLEX WIRELESS NETWORKS

Master of Science (M.Sc) Thesis in Telecommunication Engineering

By

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# ABSTRACT

Access Control in Full-Duplex Wireless Networks

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Single-channel full-duplexing is a new concept that enables wireless stations to transmit and receive simultaneously over the same frequency channel. This concept has recently found practical aspects with the introduction of innovative antenna design approaches and implementation of signal processing methods that finally led to the fabrication of few prototypes. As has been anticipated by the theory, the empirical results have confirmed that full-duplexing (FD) can indeed enhance the network throughput and capacity as well as combat the hidden terminal problem. Moreover, the biggest potential of the FD technology would be in the area of cognitive radio networks where it enables the cognitive stations to sense (receive) and transmit concurrently without needing to dedicate separate sensing intervals to serve this purpose. This can not only reduce the vulnerability of primary users (PUs) to the interference that secondary users(SUs) can generate but it also allows the SUs to increase their channel utilization and throughput.

This fundamental change reveals the need of discovering suitable higher-layer protocols and mechanisms in the protocol stack, at the very bottom of which the FD technology is laid. In particular, the medium access layer (MAC) should be adapted properly to efficiently distribute the resources offered by the underneath layer and handle the FD related tasks that do not exist in the standard half duplexing (HD) mode. Moreover, in order to extend the advantages promised by FD to the network level, building well-tailored routing protocols that can take the most out of this powerful technology is essential, as most of the these protocols have been designed for the HD mode of operation. Certainly, the medium access layer must be the cornerstone of these efforts. In order to fill in these existing gaps, the contribution of this thesis is structured in two parts.

In the first part, we evaluate the performance of SUs in a cognitive setting when the cognitive nodes are enabled with FD technology. To that end, we compare the performance of such a system over an HD system and we demonstrate the superiority of the former over the latter. Even though the exploitation of FD improves the bandwidth efficiency and allows SUs to discover the transmission opportunities more quickly, a support from higher layers is needed. To this end, we propose progressive communication by the implementation of packet fragmentation at the MAC layer. In particular, we show that by dividing the packet into smaller, but independent, segments, the system performance, in terms of key performance metrics such as successful transmission probability and system reliability, get improved considerably. As the first study to unify the FD and packet fragmentation in cognitive radio networks, we compare the performance of FD, HD and fragmentation-enabled FD, and also identify the conditions under which the proposed method is superior.

In the second part, we propose a MAC protocol that leverages the advantages of FD by considering the issues that have been left unaddressed in previous studies. This protocol, which works based on the concepts of worm-hole routing and uncontended access in distributed access settings, can combat the hidden terminal problem in the network. Moreover, the proposed protocol is capable of forwarding the packets in a route for an arbitrary number of hops, which is an efficient approach to take full advantage of the opportunities that the single-channel FD opens up for having higher spectral efficiency.

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# TABLE OF CONTENTS

# LIST OF TABLES

**Table** **Page**

# LIST OF FIGURES

# CHAPTER 1
# INTRODUCTION

While the amount of data traffic in wireless systems is developing globally, it is significant to find new solutions in utilizing the wireless spectrum more efficiently. Right now, bidirectional transmission is only possible through time or frequency division duplexing, meaning that the transmission and reception must be done at different times or over different frequencies. Although Single-channel full-duplexing where a transceiver is able to transmit and receive at the same time and over the same frequency channel is yet in its infancy period, it can be a good solution for the high traffic future of telecommunication networks.

Full-duplex (FD) technology has big promises for the telecommunication standardization bodies, industries and research communities that believe in the cognitive radio as the prominent technology of tomorrow. FD has long been an engineering and scientific problem and no practical design for FD system has been proposed until recently with the fundamental progresses evidenced in hardware domain and signal processing techniques.

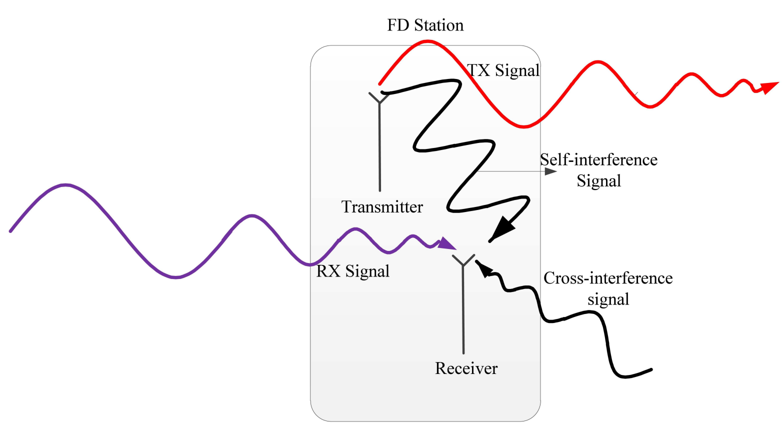



Figure 1.1: Self-interference in a full-duplex transmitter.

In fact, the reason behind all the early failures in the real implementation of this technology is principally tied to the problem of "self-interference." Technically, the difficulty in dealing with self-interference, which is caused by the infliction of interference from the transmitter of an FD station to its own receiver (Fig. 1.1), is not even comparable with cross-interference caused by other nodes, due to the immensity of its magnitude. This is because of the fact that the amount of interference is classically known to be inversely proportional to some exponent (greater than 2 according to the Friis law of attenuation) of the distance between the receiver and the transmitter. Since the FD station's transmitter and receiver are placed in close distance to each other, the RX signal is going to be completely submerged under the station's own transmitter interfering power, as shown in Fig. 1.1.

To tackle the self-interference problem, the classic solution is to turn off the station's transmitter when its receiver is on, and vice versa. This mode of operation, termed half-duplexing (HD), though it decreases the spectral efficiency and transmission throughput, has been the only practical possibility as it avoids the self-interference problem by sharing the time to alternating transmit-receive subintervals. As opposed to this approach, early implementations of the FD technology enabled the simultaneous sending and receiving at a station but over different frequency channels. Recent advancements made it possible to realize highly efficient and strong self-interference cancellation techniques in the analog, digital and hardware domains, which finally led to the fabrication of a successful prototype of FD enabled radios [3]. With this fundamental approach, FD over the same frequency channel (as opposed to different channels) has become possible.

An efficient realization of FD technology can bring chief advantages to communication systems as follows [8]:

- Capacity Enhancement: The spectral resources in the FD transceiver are fully exploited both in time and frequency, and this brings the system a doubled capacity enhancement compared to HD.

- Providing the opportunity of proposing new channel access mechanisms: The ability of transmitting a data packet and at the same time listening to the channel in FD transmitter increases the transmission probability and packet collision detection probability. This can also be a chance for proposing new channel access mechanisms that are more efficient than the currently available protocols used for HD.

- Delay and throughput improvement: FD technology equips the transmitter with the capability of overlapping the data and control packets' transmission and reception periods, thus reducing the transmission delay while enhancing the node throughput.

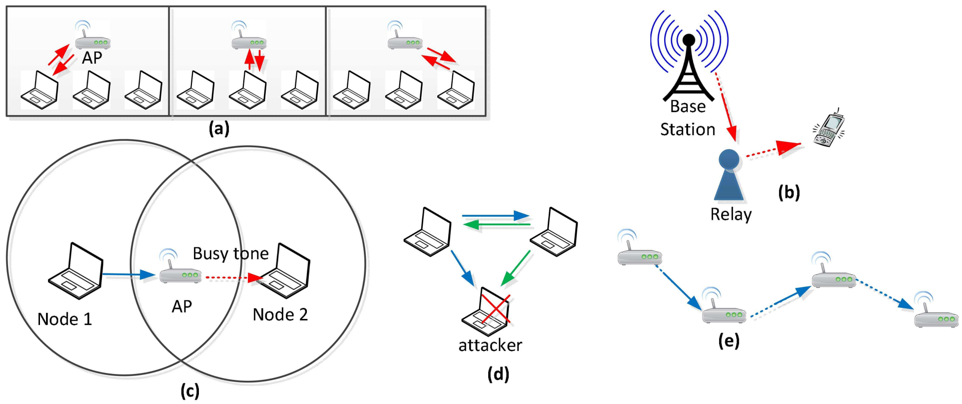



Figure 1.2: Different advantages of the FD technology.

- Network fairness progress: The network fairness can be improved in many types of FD networks such as centralized networks. In HD centralized networks with $n$ subscribers, all nodes including the access point (AP) get the same share of the channel, which is equal to $\frac{1}{n+1}$. However, since the data load of AP is $n$-times more than a regular node, the channel fairness is not the same for all nodes. Using FD technology, the AP is able to transmit at all the times concurrent with the transmission from other nodes, as illustrated in Fig. 1.2(a).

- New relaying strategies: The FD technology transforms the shape of the cooperative communication by empowering relay nodes to start forwarding a packet upon receipt of the header without needing to wait for the reception of that packet to be completed first. This brings a magnificent amount of advantages to the current and future relaying strategies. (Fig. 1.2(b) and (e)) resolves many existing problems and challenges.

- Lessening the hidden terminal problem: As mentioned before, FD enables the wireless nodes to transmit and receive at the same time. The hidden terminal problem as depicted in Fig. 1.2(c) is a negative topological by-product of almost all multi-hop communication scenarios that sometimes becomes the dominant source of error and packet loss in the network. The problem arises when the receiver station (AP in the figure) is exposed to another station transmission (Node 2) in addition to the the transmitter station's (Node 1) signal. Using the FD technology, the receiver can signal a constant busy tone (or any other data packet) which informs other nodes in the network of the current busy channel status, which ultimately prevents the hidden terminal downsides from harming the communication and diminishes the collision probability in turn.

- Security improvement: From the point of an attacker view who wants to eavesdrop two FD nodes' transmissions, at each instance of time, he/she encounters a scrambled signal that is a blend of two different signals on a single channel, which cannot easily be separated with no side information. As a result, transmission security is naturally enhanced as illustrated in Fig. 1.2(d).

Having summarized the main advantages of the FD technology, in the next chapter, we present a background on the cognitive radio (CR) and medium access layer. Then in chapter 3, we provide a literature review on the physical and MAC layers of FD systems and we show how researchers could overcome the self-interference problem and turn the dream of having FD transceiver into reality. This thesis focus being realizing the benefits that FD technology can bring to the regular wireless systems, in chapter 4 we show how equipping the network nodes with FD transceivers and applying a suitable channel access mechanism can decrease the hidden terminal problem and increase the opportunity of establishing successful transmissions. We also demonstrate that this characteristic of the FD technology plus the ability of having simultaneous transmissions and receptions can make a noticeable improvement in the capacity, throughput and delay in the considered networks. Subsequently, we study a FD cognitive radio network in chapter 4 and apply packet fragmentation in the MAC layer of the secondary user to reduce the packet drop rate and the primary's interference time while improving the energy efficiency at the same time. Finally, the thesis is concluded in the last chapter.

# CHAPTER 2
# BACKGROUND INFORMATION

At the beginning of this chapter, we give a brief review on cognitive radio networks and their famous MAC protocols (section 2.1). Then, we review different types of MAC protocols in wireless local area network (WLAN) and the main methods used in the performance analysis of these protocols (section 2.2). At the end, we explain the IEEE 802.11 distributed coordination function (DFC) and Bianchi Markov modeling in more detail. Since, in section 5.1, we use the Markov chain as the modeling tool for our proposed FD MAC and our model has similarities with the main model proposed for the CSMA/CA about a decade ago, we explain this latter model for a better understanding of what comes in later chapters.

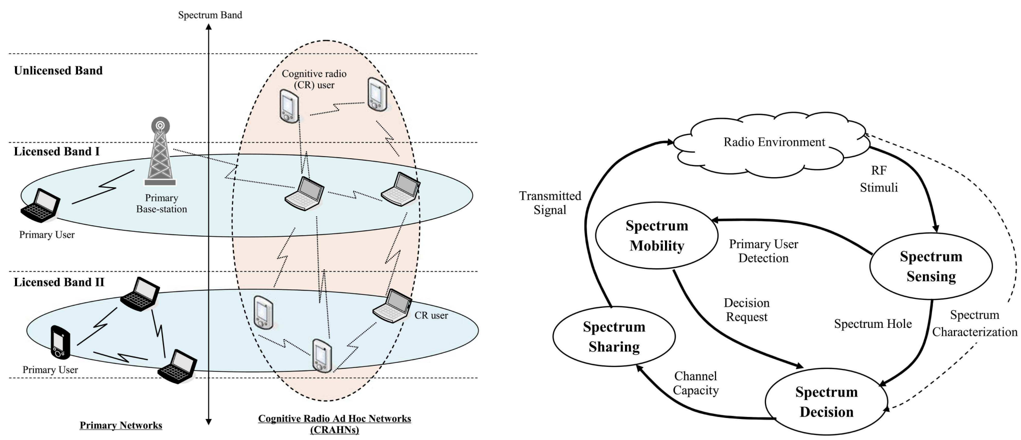
## 2.1 Cognitive Radio Networks

With the continuously increasing demands for high-bandwidth wireless applications, the current rigid frequency allocation will undoubtedly fail to meet the future needs. However, thanks to the measurements made, which showed that large fractions of the spectrum are not efficiently utilized [30], cognitive radio (CR) is paving the way for an efficient access to the spectrum holes when identified by the cognitive users.

### 2.1.1 Functional Blocks

The term CR [30], [31] refers in essence to the technology that enables a wireless station to adapt itself to the changing wireless environment. This definition simply attributes two key features [31], [32] to the CR technology, meaning vigilance and adaptability. The wireless vigilance demands that CR-enabled stations be able to discover the characteristics of the wireless environment such as the available spectrum holes, the interference level, etc., through a combination of techniques at the center of which the sensing is located. On the other hand, the adaptability is the ability of the wireless stations to constantly adapt their transmission characteristics effectively using the information gathered by the vigilance feature. These transmission characteristics can span anything from the modulation type and coding rate, to the center frequency, transmit power, antenna directivity and so forth and therefore is a general notion [33]. In a finer classification [34, 31, 35], four sets of tasks are defined for a cognitive platform: (1) Spectrum sensing (2) Spectrum decision, (3) Spectrum sharing and (4) Spectrum mobility.

It should be noted that while this spectrum-centric definition of CR with the classification given above does not govern the general definition given before, it is sufficient to solve the problem of inefficient allocation of the spectrum explained before. These functionalities are shown as modules below in Fig. 2.1 and explained one by one in the following subsections.




(a) Cognitive1a　　(b) Cognitive1b

Figure 2.1: The CRAHN architecture and the cognitive radio cycle are shown in (a) and (b), respectively [5].

#### 2.1.1.1 Spectrum Sensing

Due to the fact that in DSA the exclusive right of access does not exist anymore, any station can occupy a wireless channel. However, for this mode of operation not to turn into chaos where the simultaneous transmissions from different stations result in an unacceptably high level of interference, a sensing mechanism is required to detect the presence/absence of the licensed (PU) transmitters. Therefore, wireless nodes transmit on a channel only if that channel is empty at that moment.

Over the course of years, different sensing mechanisms have been proposed with their deficiencies and strength points. Perhaps the most common technique among all is the energy detection method where the receiver intermittently collects samples from the receiver antenna and averages them for a period of time (sensing interval), and compares the result with a pre-specified threshold to infer whether a signal is on the air or the channel is empty. It is well-known that the energy detection method is easy in implementation though it is mixed with uncertainties and performance challenges rendering it inefficient in some situations. Later on, other reliable techniques were introduced that induced more complexity into the system but outputted a more reliable decision. Just to name a few, we can mention the cyclostationary detection method and the matched filter detection method.

The important point to take into account here is that the recurrent behavior of PUs on channels, which is prompted due to their exclusive access right, calls for the periodic sensing of the channel or channels that the SU is willing to opportunistically get access to. Defining the sensing period as the interval between launching two sequential sensings over the channel(s), the first things crossing one's mind is that the sensing period shall neither be very short nor very long, as it causes a high chance of interference or it wastes the resources, respectively.

The sensing reliability, which plays an imperative role in the amount of interference inflicted to PUs, can be increased through other methods rather than decreasing the sensing period and increasing the sensing interval. One of these methods is called cooperative sensing. In cooperative sensing, stations that can either be organized in centralized or decentralized settings collaboratively conduct the sensing task and share the sensing decision outcomes with each other. Without going to much in the details, one should just bear in mind that cooperative sensing improves the sensing reliability by some factors depending on the location and the number of the collaborators and that it follows the same logic as in cooperative communication (i.e., cooperative diversity).

#### 2.1.1.2 Spectrum Decision and Spectrum Sharing

After the sensing mission is carried out, stations should be able to transmit their traffic. For the case of single-channel CRN, the determination of the presence or absence of the PU simply boils down to a binary decision made by SUs on whether they can transmit on that channel or not. However, the single-channel scenario is only a simplification of the notion of CR since CR is supposed to help in collapsing all the existing rigid boundaries, which can span several Gigahertz of spectrum. Therefore, as an ultimate target, wireless stations ought to be able to monitor a wide width of the spectrum to discover all the holes and decide between them. This decision shall be optimal considering many factors from the activity pattern of PUs to the changing wireless conditions. For example a rational choice for a SU would be to choose bandwidth for transmission that has the least PU activity pattern and has experienced the least number of deep fadings (largest SNR). In a more advanced level, such a decision can even be made in a global manner so that the network optimality is achieved by asking stations to distribute the available pools of resources in such a way, e.g., the network throughput is maximized. Therefore, the problem of allocating the resources among nodes is actually transformed to an optimization problem where the constraints are set, and the objective is to redistribute the available resources among the stations in a locality in such a way that the maximum payoff (minimum cost) is achieved. In fact many game-theoretic and optimization studies have targeted this problem recently and many interesting results were obtained. This is however beyond the scope of this thesis.

#### 2.1.1.3 Spectrum Mobility

Due to the non-exclusive access of SUs to the spectrum and, conversely, the exclusive right of PUs to access the spectrum whenever they have a traffic for dissemination, SUs are exposed to unpredictable interruptions of connections and collisions caused by the recurrence of PUs over the channel. As we introduced just before, the intermittent sensing by a SU helps the latter maintain its vigilance of the semi-instantaneous situations on the channel.

The spectrum mobility deals with the ability of SUs to keep up the quality of their logical link after the recurrence of PU to that channel is assured. Apparently, a handoff mechanism is required in order to shift the communication link to another physical channel. This indicates that a list of backup channels is necessary to be kept all the time and frequently gets updated by all the SUs for such situations. Spectrum handoff usually incurs a loss, which its severity depends on the agility of the handoff mechanism and some other factors.

In a different viewpoint, likewise other networks where the centralization of access is a key factor in classifying the emerging technology, the CRNs can be of two types [34]: (i) Infrastructure-based CRNs and (ii) CR ad-hoc networks (CRAHNs).

As its name implies, in infrastructure-based CRNs, a central entity generally called base station (BS), is in charge of the aforementioned cognitive tasks. Since the execution of the tasks that a cognitive node needs to carry out is enormously large, a central element such as an AP with presumably unlimited processing and communication power sounds more feasible in many applications. Nevertheless, akin to other networks, the centralized approach suffers from a lack of scalability and robustness, and thus can't be the only conclusive option. On the contrary, in the decentralized CRAHNs, no central element exists and stations are responsible for the aforementioned cognitive tasks in self-autonomous or cooperative manners.

### 2.1.2 Traffic Modeling

There is no doubt that the very precise determination of the traffic pattern (or activity pattern) of PUs over a channel is extremely difficult and a cumbersome task and that it can not be done in a modeling framework. In fact, in actuality, PUs represent digital TV operators and wireless microphone users, etc., with different traffic patterns and transmit powers. In rare situations where some PUs' activities follow a deterministic pattern, SUs can exactly determine the presence/absence time of PUs. However, in most of the cases, the behavior of PUs is unpredictable and seen like a random process from SUs' perspective. Hence, statistical metrics are the only tools that can be used to characterize and capture their behavioral trends. Such statistical metrics, which are obtained by time-averaging of many empirical observations, are mainly abstracted in the form of average and variance and sometimes higher moments. For the problem at hand, it turned out that average activity pattern is an important indicator reflecting the chief statistical properties of the PUs. This activity factor simply reflects what fraction of a time interval is in average occupied by a PU. The larger this figure is, the more occupied the channel is perceived to be.

The mathematical logic behind the representation of the traffic pattern using a single number is shown in Fig. 2.2 where the activity pattern of a PU is shown by a two-state birth-death process. As illustrated in this figure, state $I$ represents the absence of PU on the channel while state $B$ shows its presence. The transitions between these two states are extended using fixed probabilities $P_0$ and $P_1$. Since the dwelling times in the states of a birth-death process are exponentially distributed, it is easy to find the steady-state probabilities of being in idle and busy states in this diagram based on the given transitions probabilities. If solved, the probability of being in a busy state would be equal to the activity factor that we introduced before.

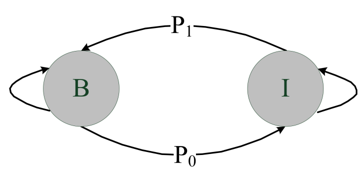



Figure 2.2: Two-state transition diagram for modeling the dynamic behavior of primary users over a channel.

To solve this diagram, the values of $P_1$ and $P_0$ are required. These values can be obtained through experimentation by sensing the channel and counting the number of times the channel goes from busy to idle and, separately, from idle to busy and dividing each count by the total number of changes (either way). Many studies took the above model as the baseline of their investigation as a well-established model [36, 37, 38, 39, 40, 41]. Therefore, when needed, we take this model for modeling the PU activities in this thesis as well.

### 2.1.3 Medium Access Layer in CRNs

Medium access layer in CRNs is of more importance compared to its counterpart in other networks due to the undertaking of the four cognitive tasks introduced briefly in the last section. Though the investigations in recent years resulted in different classifications of MAC protocols suitable for different situations, the classification by the centralization/decentralization of data and sensing planes is of more importance; for example, a combination of CRAHN with a centralized sensing scheduling, where a central element exists to optimally choose the sensors and sensing parameters, makes a perfect sense for a reliable PU protection. The choice of sensing parameters such as the sensing periodicity, sensing length, number of sensors, sensing threshold, sensing type, etc. results in totally different MAC protocols. Here we start by introducing the existing cognitive MAC protocols as follows.

#### 2.1.3.1 Random Access CR MAC

Due to the fact that the contribution of this thesis is on the medium access in FD enabled radios, it is mandatory to give an introduction on the existing major classes of MAC protocols. However, since the target network in this thesis can be cognitive-enabled or not, we start with the introduction of the cognitive MAC protocols, as this will govern the classical MAC protocols as well.

- Single Interface: Decentralized or random access MAC protocols are suitable for networks where no infrastructure exists and synchronized operation can't be achieved. Due to its popularity and its well-adopted functionality, many of the random access CR MACs are built upon the CSMA/CA access mechanism with enhanced capabilities to be adaptable for CR operation. The self-autonomous behavior of a station liberates it from any reliance on external aids and the station is responsible for all the decisions. For example, one of the important decisions of a station in the decentralized mode is to figure out what portion of the resources (i.e., power, bandwidth, etc.) should be allocated for the sensing and what proportion should go for the data transmission. This problem was proposed in [42] with the name of hardware-constrained MAC (HC-MAC) by obtaining the optimal length of the sensing interval. In this protocol, a common control channel (CCC) is dedicated for channel contentions after which the sensing at either ends of a link is initiated. Once the list of feasible channels at two CR nodes situated at the ends of a link is determined, this list is distributed in the form of a control massage the loss of which is a challenging problem by itself. In the single-radio adaptive channel (SRAC) MAC protocol [43], the presence of jammers in the CR frequencies were considered.

- Multiple Interfaces: When there is the possibility of equipping stations in the CRN with multiple interfaces, a more reliable communication can be expected. In fact, the two chief problems that substantially deteriorate the performance of random access schemes are the hidden and exposed terminal problems [44, 45]. By exploiting several interfaces, these problems can be alleviated to a large extent. For example, in dynamic open spectrum sharing (DOSS) MAC protocol [46], stations are equipped with three transceiver interfaces for control, busy-tone and data transmission purposes. The protocol is designed in such a way that a mapping exists between the data channels and the busy-tone channel, meaning that when a station obtains access to a channel (say ch. $i$), a busy tone is emitted on a corresponding frequency in a busy-tone band as well in order to notify all other nodes of the occupancy situation. Therefore, the serious problem of missing the control channel associated with [42] can be avoided. Moreover, the existence of a busy-tone band allows the other nodes to more efficiently perform the sensing tasks by only sensing the channels that do not have a triggered busy-tone signal (flag) in the corresponding busy-tone band. Therefore, transmissions from SUs are not mistaken for those of PUs. Nonetheless, it is obvious that the installation of three transceivers and dedicated control and busy-tone channels are among the incurred costs of this scheme (protocol).

#### 2.1.3.2 Time-Slotted MAC

These MAC protocols require network-wide synchronization where the time axis is split into time slots for the control and data channels [47, 48]. For instance, in the cognitive MAC (C-MAC) protocol [47], which has much similarity with IEEE 802.22, distinct slots are allocated in the beacon period to each cognitive node.

Taking the famous C-MAC as the pivot of discussion, this protocol determines the best existing channel according to the information in the beacon, named the rendezvous channel (RC), and a list of backup channels (BCs) that may be used as in reserve. The RC is then used as the control channel, which is different from the data transmission channels. Likewise for IEEE 802.22, the transmission units on the data transmission channels are called superframes comprising three major sub-periods named data transfer period (DTP), beacon period (BP) and quiet period (QP). The BP and QP are not necessary synchronized for different channels. This permits the other cognitive stations to distribute their local information using beacons in addition to enabling truthful sensing operations in the QPs. The cognitive nodes intermittently check RC to get the local information about neighbors, do the synchronization and declare any state-change in the spectrum.

#### 2.1.3.3 Hybrid MAC

The hybrid MAC protocols are designed to bring together the advantages associated with the decentralized and centralized access methods. This means that though the control signaling takes the centralized approach through the transmission over the synchronized time slots, the data transmission is done in a random access fashion or vice versa.

Among many proposed protocols, we can name [49, 50, 41]. In Cog-Mesh [49], the cognitive nodes are bundled into clusters where two types of communications are required: intra-cluster and inter-cluster. In intra-cluster communication, the beacon packet exchange, the neighbor list maintaining, and the data transmission are all carried out by the cluster heads. In inter-cluster communication, the cluster heads accept new CRs soliciting to join the network as well as undertake the data routing among themselves. A similar approach was taken in opportunistic spectrum MAC (OS-MAC) protocol [50]. However, the problem associated with any clustering-based MAC architecture is directly related to the overhead of forming the clusters and maintaining the connectivity in the presence of station mobility.

In the partially observable Markov decision process (POMDP) MAC protocol proposed in [41], cognitive nodes perform both the data transmission and the sensing task within a time slot. More interestingly, the data transmission mode is the same as the RTS-CTS scheme in the CSMA/CA access mechanism. The point of departure of this protocol pertains to the learning mechanism that assigns different weights to channels. This gets accomplished by successively assigning awards (weights) to channels that just carried a successful transmission. The protocol also optimizes the length of sensing time in each slot, which directly enhances the network throughput. There are some problems associated with this protocol as well, such as its reliance on the assumption of constant PU arrival rate (time invariant pattern), inefficient start-up phase, etc.

## 2.2 Wireless MAC Protocols

Simply speaking, the main task of the medium access protocol in the MAC layer is to schedule the network access of stations. This includes instructing nodes when to access the shared medium and using how much of the resource and in what way to do it. The channel access assignment must be in a way that the limited network resources could be fairly and effectively shared among stations [51]. In fact, designing a suitable MAC protocol according to the characteristics of the network and also its physical specifications is important in achieving the highest system performance.

According to [52], one general classification of MAC protocols is: random access, guaranteed access and hybrid access. The random access protocols such as ALOHA [53], Slotted ALOHA [54], CSMA (non-persistent, p-persistent and 1-persistent) [55] and CSMA/CA perform based on wide network contention between stations without the need for a BS to share the channel access opportunity among the stations. Thus, the back-off strategy in such protocols is essential and helps the system avoid collisions. In fact the scheduling in these protocols is obtained by the randomization of access instants.

On the other hand, in guaranteed access, the channel access assignment is done either by polling or token exchange. The polling protocols work in the master-slave form where the master node polls the slaves and in return the slave nodes send their data to the master. In the token-passing protocols, a token packet passes among the nodes and only the node that has the token is authorized to transmit its data to the BS. After the data transmission, the token will be send to the next station.

Finally, the hybrid access mechanism is a combination of the two above-mentioned methods to take advantage of both [52]. Here, using the random access mechanism, stations solicit channel access by sending access requests to the BS or AP (decentralized soliciting). When the AP receives such requests, it decides either to accept or refuse them. Upon acceptance, the BS assigns time slots to the corresponding stations and sends control messages back to these stations providing each of them with information about its assigned time slot (centralized access). Once the time slots assigned to a group of nodes, the following transmissions would be collision-free. This protocol is usually used in multimedia applications since it provides a high level of Quality of Service (QOS).

Speaking of random access protocols, we should also mention that these protocols are flexible and robust in nature and because of that, they can be reliably exploited in distributed access networks or in the hybrid schemes. In addition, random access protocols can easily incorporate some mechanisms to provide prioritized channel access to different stations [56], which is indispensable for a network's service differentiation.

While the suitable MAC protocol can guarantee high performance of the wireless network, it's important to fully understand the characteristics of the MAC protocol we are going to use in a system. In this thesis, our focus is on the CSMA/CA protocol. To that end, in the next section, we give a brief explanation on different existing models that emerged to analyze the performance of the distributed MAC protocols.

The focus is only on distributed MAC due to the fact that the MAC protocols that we will introduce in Chapter 4 and 5 for FD enabled radios are of distributed nature as well.

### 2.2.1 Analytical Models for Distributed MAC Protocols

The performance of MAC protocols can either be analyzed using simulations or analytical models. However, the problems associated with the simulation approach are time-consuming and its limited validity to the set of parameters under investigation. On the other hand, the analytical models can help us take a deeper look into the features of the protocol as they are not merely valid for a single set of parameters (and if they are, the model is not good and should be revisited and reinvented.)

Concentrating on the analytical models, in general there are three approaches taken to evaluate the performance of MAC protocols. All three are basically stochastic models but with different statements and approximations. These protocols are briefly explained below.

#### 2.2.1.1 $S - G$ Analysis

From 1970 to 1990, this model was the main approach of analyzing the performance of slotted and non-slotted protocols such as ALOHA and CSMA [55, 57, 58, 59, 60, 61, 62]. In this approach, $S$ stands for the carried load and $G$ represents the offered load [63]. Here the chief assumption is having an infinite number of nodes producing a traffic alike an independent Poisson source with a cumulative rate of $S$ packets per slot. Also, the aggregate transmissions and retransmission traffic are modeled by a Poisson process with the transmission rate of $G$ packets per slot [59]. In fact, it was proven later that the limitation of this model is related to the crude assumption of having an infinite number of nodes and a one-word size buffer for each station. Moreover, this model is more applicable to homogenous networks.

#### 2.2.1.2 Equilibrium Point Analysis

The performance evaluation of the MAC protocols gets complicated in the Markovian models especially when the number of working states increases (e.g., when the back-off stages and queue length both increase at the same time). However, by using an equilibrium point analysis (EPA), any complex steady state Markov chain can easily be approximated. As apparent from its name, the EPA is a model for the steady state operating point of a system that uses the fluid-type approximation [59, 64]. In this model, the main assumption is that the system is always at equilibrium. In other words, the traffic entering into a state at the equilibrium point is equal to the traffic leaving that state. Thus, we can find a set of nonlinear (mostly) equations whose solution will give us the value of the equilibrium point. The main advantages of this protocol is that it is not necessary to calculate the state transition probabilities of the Markov chain and that is the reason why this approach can be used in analysis of complicated MAC protocols such as the packet reservation multiple access (PRMA) protocol.

#### 2.2.1.3 Markov Analysis

Perhaps, the Markov chain is, by far, the most prominent tool for the analysis of MAC protocols. This tool is used in two main directions. The first direction is to use a Markov chain to model the states of the system. In [65] and [66], a MAC protocol has been considered for a homogenous network with the following states:

- State I - *backlogged state*: in which the frame should wait in the buffer for its turn to be transmitted.

- State II - *thinking state*: with no frame waiting in the buffer and based on the Bernoulli distribution, a frame is generated and added to the buffer later on.

In this model, a one-dimension Markov chain is used where the number of backlogged stations represents each state of the chain and obviously the number of states is equal to the number of stations. This is the simplest form of Markov analysis for MAC protocols. The multi-dimensional Markov chains are used for the more intricate protocols. For example, for the case of multi-stage back-off mechanisms, each state might represent the number of stations in a specific back-off stage [67]. Another example is the modeling of the network with multiple classes of stations. In this situation, each dimension of the Markov chain represents the number of nodes in each class (e.g., [68] for integrated voice and data system with PRMA [69]).

A different well-known approach to be taken for the analysis of networks is to use Markov chains for the modeling of the buffer status in each node, a good example of which is [63]. Nevertheless, this exploitation of Markov chains for large buffer sizes is very tedious and practically works for situations with small buffer sizes. Up to this point, the usage of Markov chains in the network level was discussed. In other words, rather than representing the number of stations, etc., states of a Markov chain might represent the states of an individual station in the network. This usage of Markov chains found popularity specifically after the studies of Bianchi in [70, 7] on the performance analysis of DCF in the IEEE 802.11 WLAN standard. In simple language, here the value of a back-off counter of a station at any instant of time represents a state of the Markov chain and the inter-state transition occurs once the back-off counter reduces or a retransmission is initiated. More details on this modeling approach will be provided in the next section.

Numerous analyses have been proposed on the modeling of the DCF mechanism grounded mainly on the Markov model of Bianchi, but entail more practical aspects of the standard (e.g., [71, 72, 73, 74, 75]). Bianchi's model was generic enough that it was even used to model other offspring protocols such as IEEE 802.11e enhanced distributed channel access (EDCA) (protocol designed to handle servicing multiple classes of traffic by prioritization) [76, 77, 78, 79], IEEE 802.15.4, which is aimed at personal wireless communication (a basis for the Zigbee protocol) [80, 81, 82, 83] and HomePlug [84, 85].[1]

Though the outputs of the Markov models introduced right before are in good conformity with the experimental data, there are some issues of precision that need to be mentioned here. In fact, one of the reasons that renders any Markov model an approximate approach is due to the memoryless assumption that is associated with it. The memoryless property in the above models attributes itself to the Poisson and Bernoulli assumptions of traffic sources and the saturation assumption (meaning the station's buffer is always non-empty). Nonetheless, even with such simplifying assumptions, the complexity of the model for a high number of states is sometimes enormous and finding a transition probability matrix and the state transition probabilities is yet a cumbersome task.

### 2.2.2 IEEE 802.11 Distributed Coordination Function MAC protocol

The MAC layer in IEEE 802.11 comes in two different flavors, DCF and point coordination function (PCF), where the PCF [86, 87] was embedded as an alternative access mechanism to handle real-time traffic using a centralized polling mechanism. The main reason for the pervasive popularity of WLANs is due to the proved success of the DCF mechanism. In fact, the popularity that DCF has obtained is not even comparable with that of PCF, which is not as efficient as its rival, has limited QoS provisioning, and bears a lot of complexity in implementation [88].

[1] HomePlug is the name for different types of power line communication technology over the house electrical wiring. The random back-off algorithm in CSMA/CA IEEE 802.11 has been used as the mechanism of accessing the channel to avoid collision between different nodes' packet transmission. [85]

#### 2.2.2.1 DCF Access Mechanism

In the DCF access mechanism, stations compete with each other to grasp the channel access. The basic rule of this mechanism is that stations must sense the channel before initiating transmission. Due to this reason, the name of this mechanism is carrier sense multiple access with collision avoidance (CSMA/CA). When a station has a packet to transmit, it first senses the channel and if there is no activity on the channel, it can initiate transmission. However, if it finds the channel busy, it will defer its transmission for a random amount of time. This random time (call it back-off) is extracted from a uniform distribution with range $[0, CW_r)$, where the value of the upper bound $CW_r$ is to be updated (in fact, increased) after each unsuccessful transmission. A station is not allowed to transmit until its back-off counter is zero. This demands that back-off counters of awaiting stations are reduced as follows: A back-off counter is reduced by one unit at the end of every idle time slot (an empty channel) and is frozen to its current value (a busy channel) for the duration of channel business. This requires that stations be vigilant of the channel status in all the time slots in order to obey this back-off decrement procedure.

The continuous sensing of a channel imposes a great deal of power and processing expense on stations, which is not desirable at all. As a matter of fact, it was proven experimentally that the amount of power that is required for sensing the channel is quantitatively alike to transmit power, a fact that renders continuous sensing an inefficient approach. In order to avoid this problem, a small twist was introduced in CSMA/CA, called the network allocation vector (NAV). Simply speaking, NAV is an information vector that is embedded in the transmitted packet and includes the remaining length of the current transmission (in bits or bytes). This vector is embedded at the beginning of ready-to-send (RTS), clear-to-send (CTS) and data packets, and simply notifies awaiting stations of the length of the time that this transmission will go on. Upon receiving the NAV, a waiting station (stations with non-zero back-off) will freeze the back-off counter and turn off its sensing module for the duration of overheard NAV and will eventually come alive once this duration is expired. This mechanism clearly helps a lot in diminishing the power consumption.

Upon expiration of NAV, the station senses the channel again; if no transmission is detected, the back-off counter decreases by one if the station finds the channel idle for the duration of DCF-interframe-space (DIFS) (usually, a few times the slot length); otherwise, the previous freezing procedure gets repeated. Since the channel activity seen from an observer's point of view is an infinite sequence of busy/idle states, the decrement procedure will reach zero at some point, which makes the station eligible to transmit.

Though the randomization obtained by the back-off mechanism diminishes the collision probability by many orders of magnitude, there still exists the possibility of collision that occurs when two or more stations finish their back-off counters at the same time and thus transmit in the same time slot. Therefore, it is mandatory that the receiver acknowledges receipt of the packet to the transmitter by a short ACK packet transmitted after a short interval, namely the short interframe space (SIFS). If the transmitter receives the ACK packet within a predefined interval ($ACK_{Timeout}$), a new contention for the next packet in the buffer will be initiated; otherwise, the station initiates the retransmission of the packet by doubling the contention window $CW_r$, extracting a back-off number, and the rest is the same as what explained so far. In the meantime, how many times a packet is allowed to be retransmitted depends on the traffic, QoS requirements and the choice of other network parameters. For a packet that is allowed to be transmitted $m$ times, if the number of retrials approaches this number, the packet will be dropped without having a chance for future retransmissions and the contention procedure for the transmission of the next packet in the buffer.

Speaking of collision, it is important to know that, though the collision event (attributed to a collision probability) in a nominal network is a rare possibility, the seriousness of this event is affected by the choice of parameters, and particularly the number of contending stations. The choice of contention window length $CW_r$ that was discussed before is one of these parameters where a larger value for this parameter makes the collision probability smaller. Normally, the initial value of a contention window is $CW_{min} = 32$ and it can increase until $CW_{min} = 1024$ (i.e., $m = 5$ retransmissions are allowed).

Finally, there are two modes of transmission defined in IEEE 802.11 DCF that differ in what comes after a node wins the contention and transmit, namely RTS/CTS mode and the basic access mode. In the RTS/CTS access mechanism, the winner of the contention transmits a control packet called RTS to the receiver before transmitting the data packet as illustrated in Fig. 2.3. Once the receiver receives this RTS packet, it will reply by transmitting back another packet called CTS. If the receiver receives the CTS packet and was able to decode it correctly, it assures that no collision has taken place and therefore the data transmission commences. Otherwise, the collision flag goes up like before, and the same contention procedure starts again. On the contrary, in the basic access mode, the contention winner transmits the data packet immediately after winning the contention and no control packet, RTS or CTS, is exchanged between pairs.

Though the exchange of RTS/CTS control packets between transmitter-receiver pairs for every single transmission incurs a large overhead, it lowers the collision window by orders of magnitude (in particular, for lengthy data packets) as the case of collision is resolved within the duration of RTS+SIF+CTS.[2]

#### 2.2.2.2 DCF Modeling

The performance of the 802.11 DCF access mechanism has been analyzed in [7] and expanded in [71] using the discrete time Markov chain (DTMC) tool. This model helps to identify the important quantities such as transmission and collision probabilities, throughput, etc. Since the proposed model for the FD MAC in chapter 5 is built having the Bianchi's DTMC as the foundation, a detailed introduction to this model will be provided in this section.

A DTMC [89] is a discrete-time stochastic process where the future state of the process depends only on the current state and is independent of the process history. Briefly, the DTMC introduced in [7] is a 2-D stochastic process $(s(t), b(t))$ in which $s(t)$ represents the retransmission stage in the DCF mechanism and $b(t)$ stands for the back-off time counter in each of these transmission stages. As is known, the back-off counter either counts down at the end of a time slot, if the channel is idle, or freezes its previous value, if the channel is busy. Nevertheless, the busy status of the channel can be either due to the successful transmission or collision from one or more nodes located somewhere in the hearing range of this node. All the waiting nodes persist in hearing the channel and once the channel goes idle again (which happens due to the completion of the preceding transmission), they unfreeze their back-off counters. Since we can't go deep into the details of access mechanism in CSMA/CA, we refer interested readers to any of the above papers to gain deeper knowledge on the topic. All these reflected as a 2-D DTMC as proposed by Bianchi in Fig. 2.4. All states in such DTMC are non-periodic and non-null, and thus the process is ergodic [89] and a stationary distribution exists.

When the back-off counter of a node reaches zero, the node is eligible to initiate a transmission immediately. If this transmission faces a collision, the packet would be refrained for the next transmission and the node starts a new back-off counting. In fact, at each new stage (retransmission trial) the back-off $b(t)$ is chosen uniformly from the interval $[0, W_i]$, where $W_i = 2^i W_0$ for $i = 0, \cdots, m$. Here $m$ is the maximum back-off stage and $W_0$ is the contention window of the node when a packet is going to be transmitted for the first time (stage zero). Considering an infinite number of

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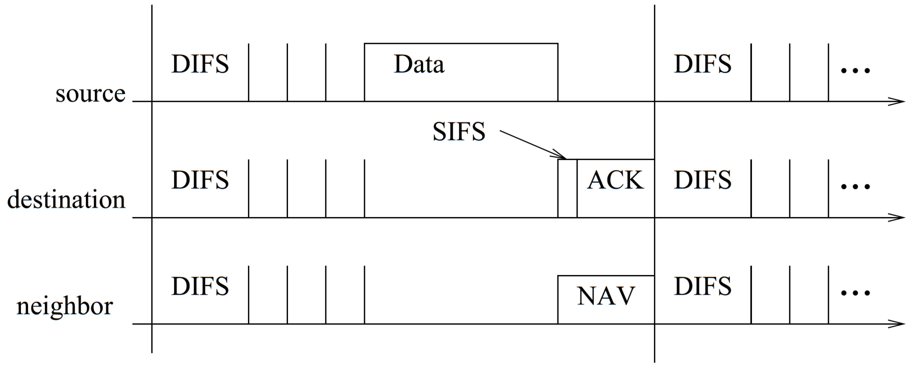
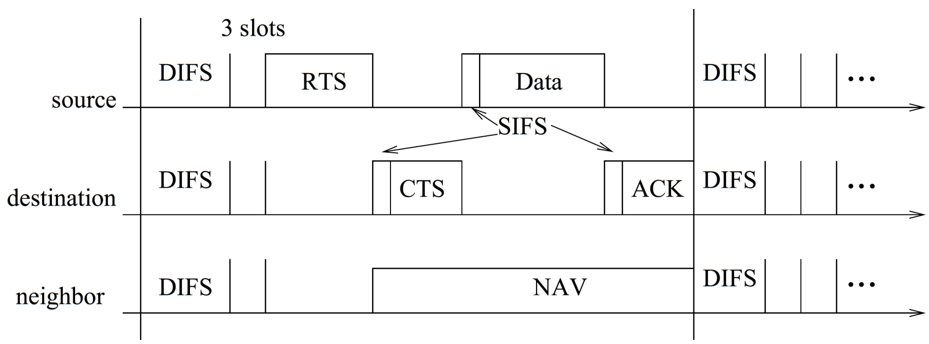
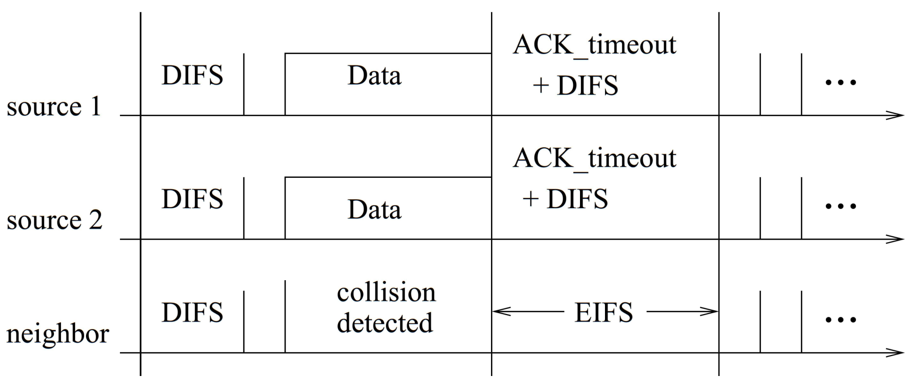
[2]This is an improvement compared to the basic transmission mode where collided transmissions are not stopped and can prolong for a long time, wasting valuable resources without involved partners knowing of the raised situation.




(a) CSMAacessa




(b) CSMAacessb




(c) CSMAacessc

Figure 2.3: The IEEE 802.11 channel access mechanism [6].

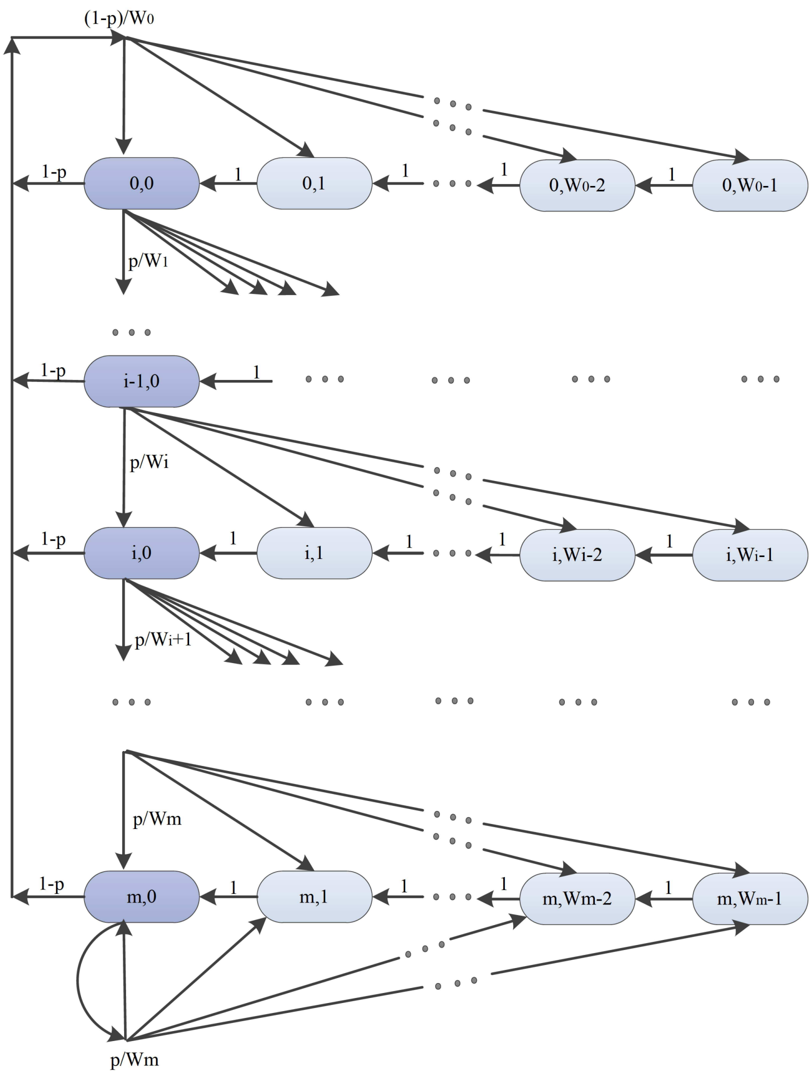



Figure 2.4: Proposed 2-D DTMC in [7] for the modeling of the DCF mechanism.

retransmissions for each packet, apparently after the $m^{th}$ retransmission the back-off length doesn't increase any more. Of course, there are some assumptions made in Bianchi's model to make the model analytically tractable; these assumptions have also been made in [90] and are as follows:

- Transmission failures can only be due to the collision of two packets.
- All nodes in the network are in the saturated state, meaning that they always have packets waiting to be transmitted.
- The probability of packet collision is constant in time and independent of the collision history, i.e., the probability of collision is totally independent of the retransmission stages of a node.
- The existence of hidden and exposed terminal problems is neglected.

Having the above-mentioned assumptions in mind, and defining $\tau$ as the probability of transmission, then one would simply discover that the probability of collision is:

$$p = 1 - (1-\tau)^{n-1}. \tag{2.1}$$

Defining $P(s(t) = i, b(t) = k)$ as the time-dependent probability density function (PDF) of being in the state (i,k) in Fig. 2.4, then its corresponding stationary distribution would be:

$$b_{i,k} = \lim_{t \to +\infty} P(s(t) = i, b(t) = k). \tag{2.2}$$

The transmission probability $\tau$ could be derived as:

$$\tau = \sum_{i=0}^{m} b_{i,0}. \tag{2.3}$$

By applying a little bit of algebra, $\tau$ could be derived as

$$\tau = \frac{b_{0,0}}{1-p} = \frac{2(1-2p)}{(1-2p)(W_0+1) + pW_0(1-(2p)^m)}. \tag{2.4}$$

In the analysis of CSMA/CA, it's important to have the numerical value of $\tau$ since it is one of the main parameters needed to calculate the throughput, delay and other QoS parameters of a node.

# CHAPTER 3

# LITERATURE REVIEW

In this chapter, we present a comprehensive review on the FD technology literatures. We start with the articles proposing solutions for handling the self-interference problem and then we study the existing FD transceivers that have been implemented so far. The proposed MAC protocols will be introduced in the latter section for different FD networks and the last part of the chapter presentees a brief summary about the main contribution of this thesis.

## 3.1 Self-Interference Cancellation Methods

As we mentioned before, combating the self-interference effect is the key challenge in the FD transceivers. In the situation that the wireless nodes are equipped with only one transmit antenna, increasing the separation distance between the transmitter and receiver to reduce the self-interference effect is the most intuitive method. However, the separation distance does not seem practical for different devices with different size and applications. Therefore, other ideas were proposed lately. Fig. 3.1 shows the physical layer of an FD transceiver using three main methods of self-interference cancellation. In fact, a review of the existing literature reveals the following methods using which self-interference cancellation is accomplished.

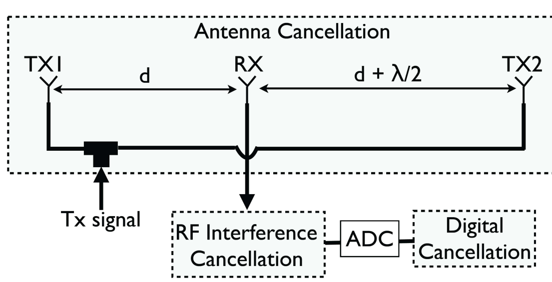



Figure 3.1: Three methods of self-interference cancellation in the FD transceiver [1].

### 3.1.1 Antenna Cancellation

The idea of the antenna cancellation method is based upon the fact that a single transmit antenna cannot considerably diminish the self-interference. As a matter of fact, the system operation in high signal-to-noise ratio (SNR) regimes requires that the system limits the amount of interference rather than dealing with eliminating it once it has already been mixed with the desired signal. This understanding helped to enlighten investigations on antenna cancellation methods where two antennas, rather than one, are used at the transmitter side. Indeed, by separating the transmit antennas and splitting the transmit power between them, it was demonstrated that the accumulative field patterns produced are added up constructively in some spots while destructively in other spots in such a way that few null spots are observed. This method is shown in Fig. 3.2. For instance, for the situation of equal split power, these null spots are located on the line connecting the transmitters, where the difference of distances to transmitters is odd multiples of half the wavelength, i.e., $\Delta d = d_2 - d_1 = (2m-1)\lambda/2, m = 1, 2, \ldots$ .

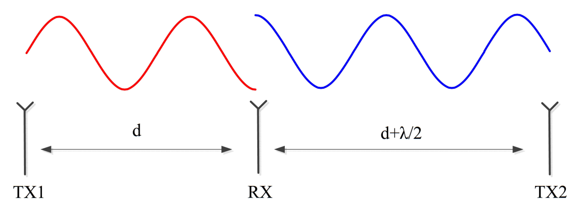



Figure 3.2: Transmitter and receiver antennas arrangement producing null point [2].

Thus, by putting the receiver at these spots, considerably less self-interference is experienced. Nevertheless, the number of null spots that exist depends on the distance between the transmitters, and if $\Delta d < \lambda/2$, no null spot exists. This immediately indicates that one of the limitations associated with the two-antenna cancellation method is the distance. Moreover, this method cannot be feasible when the operating frequency is low ($\lambda$ is high) since the device size will be a bottleneck. Further to above limitations, this technique is not suitable for wideband signals, as they are composed of many frequency components and thus a null region is required rather than a null spot, if an even cancellation is sought. The other drawback of this approach is the creation of many null spots in the far field, which is not desirable at all. Lastly, as shown in [9], antenna cancellation requires manual tuning, which can't be an ultimate solution for exploitation in highly varying wireless environments with shadowing, fading, etc. An alternative approach to alleviate these problems would be to use more than three transmit antennas at the cost of larger implementation complexity [9].

Hence, the antenna cancellation with three antennas (two transmitters + one receiver) is defeated by the three-antenna multiple-input multiple-output (MIMO) technology [10], as the former doubles throughput while the latter triples the throughput. In [11], the antenna cancellation method was refined to eliminate the far-field null zones. Using a *Balun* circuit, which creates a $\pi$ degree phase shift, makes this method independent of the signal wavelength, which comes at the cost of higher hardware design complexity. By implementation of a two-level cancellation, their pioneering design in MIMO FD system is obtained. However, this scheme is only suitable for a communication range less than 3 meters, as evidenced in [11].

### 3.1.2 Digital Cancellation

This self-interference cancellation method was first used in [12] and then it has been used in [9, 13, 14, 15, 16, 17] . In general, further to its application in FD, this approach has found profound application in optical networks and carrier sense multiple access with collision notification (CSMA/CN) as well. The main idea in this technique is to subtract the fed-back transmitted digital symbols (self-interference symbols) from the base-band received symbols in the digital domain so that a clearer version of the signal is decoded [18, 19, 10].

Traditionally, this method was used to detect the desirable signal collided with a transmission from another node. This was accomplished by detecting the undesirable signal, re-modulating it, and then subtracting it from the mixed received signal after the analog-to-digital conversion (ADC). In the case of FD, the complication is less as the transmitted symbols are perfectly known in the transmitter and, thus, de-modulation and decoding are not required. Therefore, by coherent detection, the received signal is correlated with the known self-interference. The difficulty in this process is the unknown delay and phase differences of received and transmitted signals, which can be obtained by correlation of these signals and then feeding the phase and delay information to the subtractor.

The block diagram of this method is shown in Fig. 3.3. The empirical results show that the digital cancellation is effective in both high and low signal-to-interference plus noise ratio (SINR) regimes and can cancel up to 20-25 dB of self-interference [15, 16]. The limitation associated with this technique is its sensitivity to the limited dynamic range of ADCs as the self-interference signals are very strong while the received signals are extremely weak making it impossible to recover the digital samples in some situations.

### 3.1.3 Analog (RF) Cancellation

In this approach, the self-interference cancellation is carried out in the analog domain on the radio frequency (RF) signal before the information is fed into the ADC. The first realization of this technique is to use a second transmission chain to construct an analog cancellation signal from a digital estimate of the self-interference [13], removing self-interference from signals with bandwidth 625 kHz for 33 dB. The other realization is similar to the noise-cancellation in headphones [14]. One way or the other, this goal can be achieved by deploying an RF cancellation chip [20] prior to feeding the ADC with the analog signal. This shows to help reduce the self-interference by 25 dB [14].

An example of this method was implemented by Radunovic et al. [14] who used a noise cancelling chip called QHx220 (Fig. 3.3) to remove the analog interference from signals operating in FD mode in 900 MHz frequency by subtracting the self-interference and the received RF signals as input and outputting the clear signal. This chip was able to accomplish this task by automatically adjusting the phase, delay, and amplitude of the internally fed interference signal to match the interference that is received in mixture with the desirable signal.

Nevertheless, the analog cancellation technique faces many complications among which is the non-linearity of the amplifiers.

### 3.1.4 Realization of Full-Duplex Technology

Probably [9] and [13], and [2] which appeared shortly after, are the pioneering studies that coined innovative analog cancellation methods (antenna cancelation and signal subtraction) for a practical realization of FD. The idea taken in these works is based on the simultaneous implementation of three cancellation techniques mentioned above. The striking results in [9] and [2] proved for the first time that a SIC gain of 70 and 73 dB can be obtained by the combination of these techniques, Fig. 3.3.

In [13], a comparison of the efficiency of the three different cancellation mechanisms was presented: (i) antenna separation and digital cancellation (ASDC), (ii) antenna separation and analog cancellation (ASAC) and (iii) antenna separation and analog/digital cancellation (ASADC). Their study is mostly similar to the Bluetooth and WiFi systems in terms of power and transmission range specifications. The results showed that ASADC has the best performance among the three methods, with an average interference cancellation level of 78 dB. The results prove that by combining these techniques with other methods, the amount of self-interference can be reduced by 70 dB, thus restricting the remaining interference within a few dBs of the noise floor. This is a substantial improvement compared to previously known

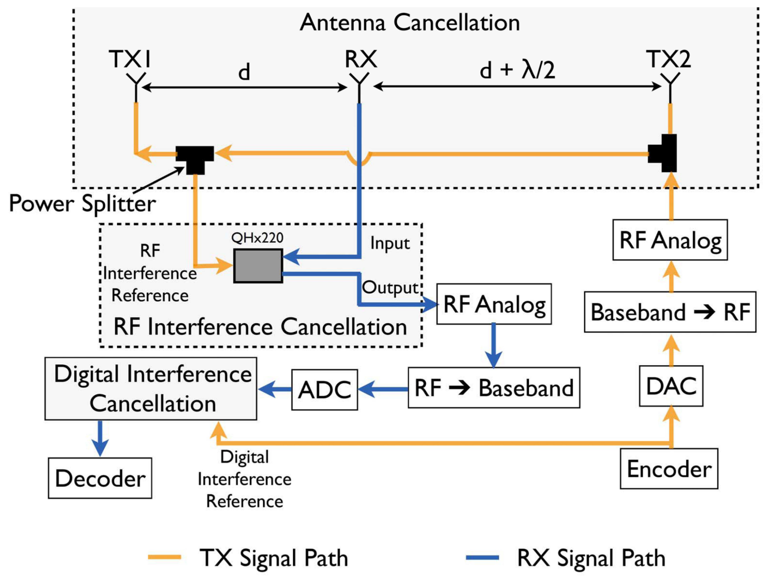


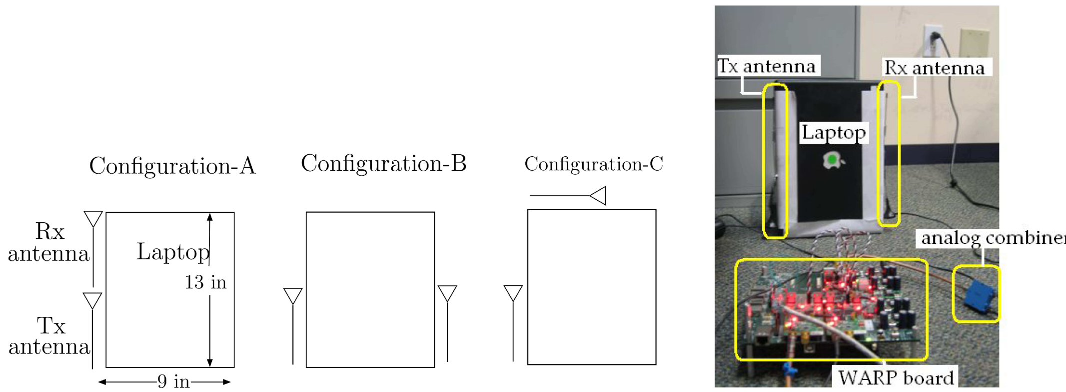

Figure 3.3: Detailed diagram of FD transceiver equipped with three methods of cancellation [2].

methods of RF and noise cancellation [20], [4], [14], and digital domain cancellation [15], [21].

In [3], the authors fabricated a practical prototype of an orthogonal frequency-division multiplexing (OFDM) FD transceiver with 64 real-time sub-carriers and MIMO antennas on a wireless open-access research platform (WRAP). In this work, different antenna placements have been tested to find the best configuration when the transceiver is installed on a mobile device such as laptop (Fig. 3.4(a)). Among these different placements, configuration-B shows the best self-interference cancellation results both in the presence and absence of analog cancellation. Since this configuration is also the best for MIMO systems, it is possible to have two modes of operation (FD and MIMO) and switch between them as needed. This design shows 80 dB of signal attenuation which is 10 dB higher than the results of previous works as the authors reported. This is the most recent and practical FD transceiver that has been designed so far (Fig. 3.4(b)).

## 3.2 Medium Access Layer of Full-Duplex Systems

The impressive progresses in FD physical layer technology rapidly unfolded the needs to investigate how higher layers of the protocol stack can support the FD




(a) Different Antenna placements (b) FD transceiver

Figure 3.4: OFDM FD transceiver and different antenna placements for this system [3].

technology. In particular, the medium access control (MAC) layer that works with the released resources and the routing layer which is supposed to lead information flow along a virtual connection are of chief importance. In order to expand the advantages promised by FD to the network level, building well-tailored protocols and mechanisms that can take the most out of this powerful physical layer (PHY) technology is essential. Certainly, the medium access layer must be the cornerstone of these efforts.

To the best of our knowledge, [2, 3, 4, 22] are the main studies that worked on the MAC protocols for this new PHY technology and inspired researchers to revisit older studies introduced back in 2006 such as [23]. In [2], authors proposed a MAC layer for centralized access networks along with presenting a method of SIC using *Balun* circuit and digital cancellation technique. The MAC protocol of this system works as is shown in Fig. 3.5: Node 1 starts transmission to AP. As soon as AP opens the header of this transmission, it transmits back to Node 1 if it has a packet to deliver to it, otherwise AP sends a busy tone till the end of Node 1's packet reception. This busy tone can make node 2 aware of the transmission that AP is involved with it and lessen the hidden terminal problem in the considered networks. In the case that AP's packet is smaller than node 1's packet, the busy tone can be signalled by AP for the rest of Node 1's transmission period to fully solve the hidden terminal problem.

FD-MAC was introduced as another protocol suited for centralized networks in [3] . It's a random access protocol that uses the concepts of the 802.11 distributed coordination function (DCF). This MAC protocol works based on three novel mechanisms:

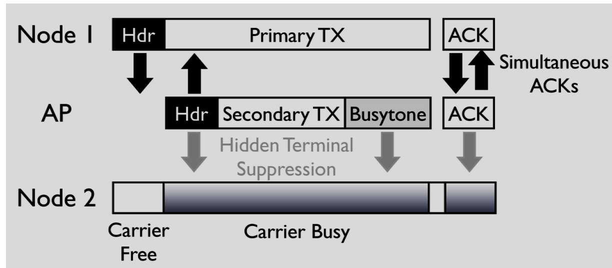



Figure 3.5: Full-duplex packet exchange of the MAC proposed in [2].

- Shared random back-off: In this mechanism, if two nodes have so much traffic to exchange, they share a back-off counter that helps them to start a synchronized FD transmission while it gives them the chance of attending in the contention along with the other nodes of the network. This method helps to increase the fairness of the network.

  Let us say that node M1 and the AP both have packages to exchange. If AP gains the channel after the contention, it starts transmitting in the HD mode to M1. When M1 receives AP's package successfully, it sends an acknowledgment packet (ACK) back to the AP with a bit called "head of line" (HOL) set to 1 in HD mode. $HOL = 1$ means that the first packet in the M1 buffer is destined for AP. At this point, both nodes know that they should switch to FD mode since both have packages to exchange. After the first FD transmission, two nodes may still have more data packets to handshake. In this case, AP and M1 pick a back-off number randomly from $[0 : CW_{max}^{AP}]$ and $[0 : CW_{max}^{M1}]$ independently, where $CW_{max}$ is the maximum contention window of each node. Then, they transmit these numbers in their ACK packages as the shared random back-off (SRB). $Max(SRB_{AP}, SRB_{M1})$ will be chosen at both nodes as the shared back-off number and both nodes start to countdown until the back-off timers expire. The only difference between SRB and a general back-off is that the countdown process for SRB doesn't pause once the channel gets busy. Instead, after the expiry of the back-off the nodes wait for DIFS, check the channel and start their next FD transmissions if the channel is not occupied. In case another node wins the contention before SRB expires, AP and M1 should go through the regular contention to access the channel for the remaining data packets.

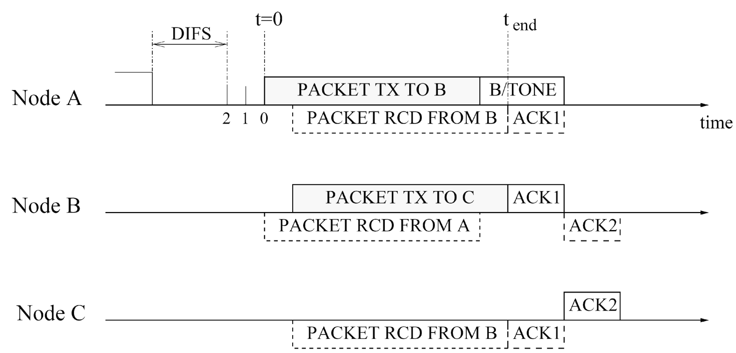
- Header snooping: In this mechanism, by snooping in the headers of all transmissions that are going on in the network, it would be easy for the node to estimate the topology of the local area around them to assign the suitable transmission strategy.

Suppose that there is a data transmission among AP and M1 in the FD centralized access network. If AP sends a packet to M1, M2 (another node in the network) has the ability to decode the header of this data packet, find its destination (here M1) and learn if the AP has another packet for that destination and willing to start the FD transmission. After AP's packet transmission finished, M2 waits to hear the ACK of the destination (M1). If M2 hears the ACK, it implies that M1 and M2 are in the same hearing range and M2 must keep quiet till the end of that transmission. However, if M2 could not hear the ACK of M1, it concludes that they are hidden from each other and updates its knowledge of network topology. By doing another snooping in the AP data packet, M2 discovers if the transmission is in FD or HD mode. If the transmission is in HD mode, M2 can set up its own packet transmission to the access point but if the transmission is in FD mode, again M2 stays idle not to cause packet collision. Although the channel estimation error due to the packet failures is noticeable in this method, still the performance of the network increases to an acceptable level by using the snooping method.

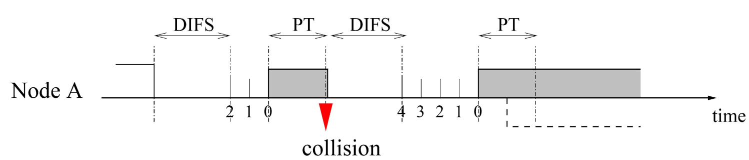
- Virtual contention resolution: Finally, in this mechanism, a statistical decision can be made by AP to serve the high priority packets earlier so that the network can take more advantages of the FD mechanism. Increasing the data buffer length can increase the number of FD transmissions between AP and other nodes in this centralized access MAC protocol. However, this may cause congestion at AP and decrease the fairness of the network. To tackle this problem, the authors proposed two different solutions that are explained in the following examples.

  First, suppose that AP and M1 have some packets available in their buffer to exchange and due to this availability an FD transmission will be established among them. In this case, reordering the M1 (AP) destined packet to the AP's (M1's) buffer head gets done with a geometric probability. This means that the probability of establishing the next FD transmission decreases geometrically, too. Second, in order to avoid collision between an FD transmission and other transmissions, a node that gets the chance of FD transmission sends its packet with the probability of $\beta/CW_{max}$, where $\beta$ is a constant related to the aggressiveness of the network. So if a node has a larger $CW_{max}$, which shows the node's higher packet collision history, the probability of getting involved in an FD transmission would be less.

For distributed networks, not too much work has appeared so far. To the best of our knowledge, [4], which proposes a MAC protocol for FD based on CSMA/CA, termed CONTRA FLOW, is the only distributed MAC protocol that has been proposed so far. In this protocol, as Fig. 3.6(a) shows, once node $A$ captures the channel according to the CSMA/CA back-off mechanism, it starts transmitting its packet to node $B$ and waits for the primary timer ($PT$) to expire. As soon as node $B$ opens the header, it starts forwarding the packet to node $C$ through an uncontended transmission. If node $A$ can hear node $B$'s packet before the timer expires, it continues the transmission. Otherwise, it stops and runs a transmission trial for another contention (Fig. 3.6(b)). Then, node $B$ forwards the data to node $C$ along with an acknowledgement packet to $A$; at this time node $A$ signals a busy tone till it hears the ACK of $B$. After node $B$'s data transmission to $C$ gets completed, node $C$ sends an ACK to $B$.




(a) Successful dual-link transmission




(b) Primary collision

Figure 3.6: Contra Flow MAC protocol data exchange format [4].

Although the idea of [4] for giving permission to a node to forward the data instantly and without going through contention is novel, it has some challenges such as high collision probability since the second hop is severely exposed to transmissions from hidden terminals. Furthermore, the corresponding MAC layer is not designed to permit more than two hops of progress, whereas two is not necessarily the optimal number of hops for uncontested progress.

## 3.3 FD in Cognitive Radio Networks

The considerable elimination of the self-interference motivated researchers to rapidly extend their investigations on the applications of FD technology in different fields of wireless communication, such as indoor mesh networks [14], multi-path routing [24] and MIMO antennas. However, the area of CR is the one that will be favored the most from the advances in this domain [25], [1] as will be discussed in this section.

The range of services that are emerging and developing to facilitate the human's life is so diverse that the classical inefficient allocation of spectrum can't keep pace satisfying them. Though the technological advancements in innovating highly efficient modulation and coding techniques has increased the bandwidth efficiency letting more data to be transmitted in a fixed amount of bandwidth, the improvements are yet inadequate. Fortified by the observation of this disharmony, the federal communication commission (FCC) realized that this problem lies at the heart of the inefficient allocation of bandwidth and therefore the dynamic spectrum allocation (DSA) was introduced with the CR as the enabling technology. In effect, in contrary to the fixed allocation strategy of resources, the main feature of DSA is the flexibility it brings in the sense that the spectrum is allocated to demanders (service provider, wireless stations, etc.) on an opportunistic basis, thus allowing a more homogeneous and efficient spectrum usage.

With the aforesaid development of FD technology, it became possible to eliminate these constraints and it got practically feasible for a wireless device to sense (receive) and transmit concurrently (using the self-interference cancellation methods) without requiring to dedicate separate sensing intervals to serve this purpose. In case the secondary user (SU) detects the presence of a primary user (PU), he releases the channel immediately, causing no harm to the primary's transmissions compared to the scenario with HD where the vulnerability interval can be as large as the inter-sensing interval. This almost doubles the channel utilization, and throughput in return, at the expense of some hardware complexity [9].

In this context, [1] proposed a FD scheme for CR networks that employs the three well-known interference cancellation methods (antenna cancelation, RF interference cancellation and digital interference cancellation), and compared the performance of FD and HD taking the PU's packet loss rate as the core factor. In [25], the authors proposed a sensing scheme for FD-enabled non-slotted CRNs and evaluated the system performance under non-ideal self-interference cancellation. For our review to be fully inclusive, we also cite [21], [26] and [27] as the studies uniquely focusing on PHY aspects of the union of FD and CR technologies, such as power allocation, outage probability and rate regions.

## 3.4 Thesis Contribution

This thesis has a two-fold contribution.

In the first part of the thesis, we aim at evaluating the performance of SUs in a cognitive setting when the cognitive nodes are enabled with the FD technology. To that end, we compare the performance of such a system over an HD system and we demonstrate the superiority of the former over the latter. Even though the exploitation of FD improves the bandwidth efficiency and allows SUs to discover the transmission opportunities more promptly, a support from higher layers is needed.[1] To this end, we propose progressive communication by the implementation of packet fragmentation at the MAC layer. In particular, we show that by dividing the packet into smaller, but independent, segments, the system performance, in terms of key performance metrics such as successful transmission probability and system reliability get improved considerably. As the first study to unify the FD and packet fragmentation in CRNs, we compare the performance of FD, HD and fragmentation-enabled FD, and also identify the conditions under which the proposed method is superior.

In the second part of this thesis, we propose a protocol that leverages the full advantages of FD at the MAC layer by considering issues in the MAC layer of FD technology that have been left unaddressed in previous studies. Particularly, our contribution in this part is an extension to the famous study [4] for increasing the number of hops that the data packet can traverse through. Decentralized (distributed) access is considered and the protocol is termed distributed-access full-duplexing MAC (DFD MAC). In the decentralized networks under consideration, the worm-hole routing technique [9], [28] is used in conjunction with uncontended access [4]. Therefore, to reduce the hidden terminal negative effect, the data header is to be sent prior to the transmission of the data packet along the path starts. This guarantees the path reservation which immediately decreases the collision rate. Moreover, the proposed protocol is capable of forwarding the packets in a route for an arbitrary number of hops, unlike the approach in [4], which is solely confined to two-hop configuration. This is an efficient approach that is designed to take full advantage of the opportunities that the single-channel FD in the PHY layer opens up for having better spectral efficiency. The access mechanism of our proposed MAC protocol implements features similar to those of the random access in CSMA/CA, such as random back-off, freezing and countdowns [29].

[1] Such higher-layer packet-level supports are particularly important in CRNs because an abrupt transmission cessation enforced by the presence of PUs could severely damage the performance of SUs since the whole packet is to be dropped all at once in order to take the most out of it and enhance the perceived quality.

In order to prove the superiority of our MAC protocol, we build a Markov model and analyze the performance of the proposed protocol using a discrete-time Markov chain (DTMC). Through mathematical derivations, we find the path throughput and end-to-end delay in the network, two quantities that should be taken into consideration instead of the single-hop counterparts. We also validate our scheme by comparing metrics such as throughput and delay with the HD CSMA/CA counterparts. Numerical results are provided and show that our protocol achieves better performance than the CSMA/CA protocol operating in HD mode.

# CHAPTER 4

# FULL-DUPLEX COGNITIVE RADIO FRAGMENTED-ENABLED SCHEME

In this chapter, a cognition scheme whereby nodes are enabled with the FD technology at the PHY layer, in conjunction with packet fragmentation at the MAC layer, is proposed to increase the performance of CR networks.

As we mentioned before, the usage of FD technology would help the SU to detect the presence of PU on the channel and decrease the cause of PU packet collision. However, even though the exploitation of FD can improve the bandwidth efficiency and allows SUs to discover the transmission opportunities more quickly, a support from higher layers is still needed. Indeed, the major problem in FD-enabled CRN is the interruption of transmission by a SU due to the appearance of a PU. This could severely damage the QoS of the secondary, especially that the whole packet should be dropped all at once. In this chapter, we propose the implementation of packet fragmentation at the MAC layer as an upper-layer support to help improve the performance for the SUs to a notable level. In particular, we show that by dividing a packet into smaller, but independent segments, the performance, in terms of key performance metrics such as successful transmission probability and system reliability get improved considerably. As the first study to unify FD and packet fragmentation in CRNs, we compare the performance of the fragmentation-enabled FD (FFD) cognition to the conventional scheme with HD and also to FD with no fragmentation, and also identify the conditions under which the proposed method is superior.

In detailing these contributions, this chapter is organized as follows: In section 4.1, after explaining the operation of the fragmentation-enabled FD network, we present the mathematical derivations of the system parameters such as average successful packet transmission time, average successfully transmitted data fraction, packet dropping time and energy efficiency. Then in Section 4.2, we discuss simulation results and see how the forenamed parameters can be improved in the proposed scheme.

## 4.1 Full-Duplex Fragmentation-Enabled Cognition

In this section, we describe the details of the proposed scheme. As mentioned before, implementing fragmentation in the MAC layer of the full-duplex CR scheme allows SUs to transmit their data in a progressive manner in a highly non-deterministic cognitive environment. In this vein, we start by giving an introduction on fragmentation at the MAC layer. Then taking the widely accepted Poisson assumption for the primary's traffic pattern, we derive parameters that directly impact the secondary's QoS such as average successful transmission time, packet drop time and average successful transmitted information. At the end, the energy efficiency of the scheme will be derived and compared with its counterpart in conventional methods. Hereafter, we consider cognition by a single SU. Indeed, our main focus being on the proposal of the aforementioned cognition scheme and the analysis of its advantages compared to conventional schemes, its application to networks with multiple SUs and corresponding analysis can be part of future extensions of this work.

### 4.1.1 Fragmentation

So far, several studies demonstrated the benefits that fragmentation can bring about in WLANs [91], [92]. For example, the results of [91] show that the optimal fragment size that minimizes the energy consumption of HD system with CSMA/CA MAC is 300 bytes for 1500 byte packet lengths, with 550 bits of overhead required on each fragment. Despite the fact that fragmentation of a payload data unit (PDU) increases the overhead, it can reduce the packet drop rate while protecting PUs from interference.

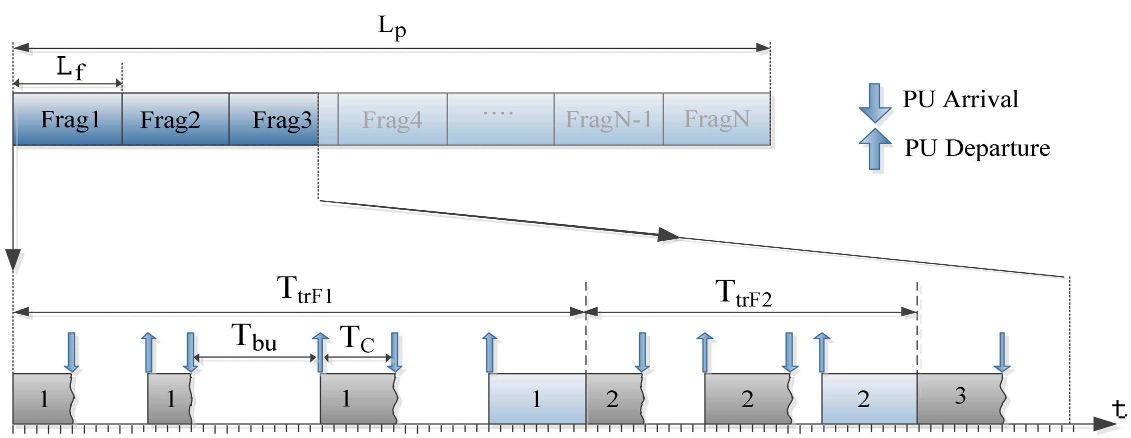



Figure 4.1: Packet fragmentation and its transmission on the channel.

Fig. 4.1 illustrates the situation where a SU data packet is fragmented into $N$ entities for transmission. Each fragment is allowed to be retransmitted at most $m$ times and $T_{trF_i}$ represents the successful transmission length of the $i^{th}$ fragment. A transmission of a fragment might be halted because of the appearance of PU(s) (downwards arrows) and re-attempted when the spectrum is clear from primary activity (upwards arrows).

Denote $L_f$ as the length of a fragment and $L_o$ as its associated overhead size, then the effective payload of each fragment is given by

$$L_{eff} = L_f - L_o. \tag{4.1}$$

Thus, one can easily define the number of fragments in a data packet as

$$N = \left\lfloor \frac{L_d}{L_{eff}} \right\rfloor + 1, \tag{4.2}$$

where $L_d$ is the total effective payload of the non-fragmented data packet and $\lfloor . \rfloor$ represents the floor function. Therefore, the overall length of the SU data packet after fragmentation would be equal to:

$$L_p = NL_f, \tag{4.3}$$

where $L_p > L_d$ because of the fragmentation overheads.

The transmission of a data packet is successful if all its fragments are successfully delivered. In fact, fragments are to be transmitted in order, meaning that the next fragment transmission would not be initiated unless the previous fragments are already delivered. Each fragment is allowed to be retransmitted up to $m$ times if the presence of PUs causes the initial transmission to be aborted, and the whole packet is dropped if a fragment cannot go through once the limit on the transmission attempts is reached. A collided fragment would be retransmitted after the PU leaves the channel. In the next parts of this section, we are going to derive the necessary parameters for evaluating the performance of the system, such as the average successful packet transmission time, $\overline{T}_{tr}$, the average packet drop time, $\overline{T}_d$, etc. However, before proceeding we describe the activity pattern of PUs.

### 4.1.2 Traffic Pattern of Primary Users

In this work, we model the PUs' arrival as a Poisson process with arrival rate $\alpha$. The probability of $k$ arrivals during the interval of $t$ units in length is

$$f(k,t) = \frac{(\alpha t)^k e^{-\alpha t}}{k!}. \tag{4.4}$$

Further, we assume that PU channel busy times are exponentially distributed with mean $\mu$:[1]

$$f(\mu,t) = \mu e^{-\mu t} \;\; t \geq 0. \tag{4.5}$$

Accordingly, we model the primary's traffic pattern with the widely accepted ON/OFF model. Furthermore, as we focus on the advantages of the proposed cognitive operation, incorporating FD with fragmentation, we ignore the expected outcome of erroneous sensing and assume that the sensing results are perfect.

Given this characterization, we are now able to define the successful transmission probability of a SU's fragment. In fact, a single transmission of a fragment is deemed successful, with probability $p_s$, if no PU appears in this time ($k = 0$ and $t = L_f/R$ in (4.4)), expressed as

$$p_s = f(k = 0, t = L_f/R) = e^{-\alpha L_f/R}, \tag{4.6}$$

where $R$ denotes the data transmission rate.

Subsequently, the probability that a fragment gets transmitted during one of its $m$ trials is given by

$$p_t = \sum_{j=0}^{m-1} p_s(1-p_s)^j. \tag{4.7}$$

### 4.1.3 Average Successful Packet Transmission Time

As aforementioned, to have a successful packet transmission, all corresponding fragments have to be successfully delivered. Hence, the average successful packet transmission time, denoted $\overline{T}_{tr}$, is simply the summation of the average successful transmission time of all fragments ($\overline{T}_{trF_i}$), i.e.,

$$\overline{T}_{tr} = \sum_{i=1}^{N} \overline{T}_{trF_i}. \tag{4.8}$$

[1] This characterization enforces that $1/\alpha > 1/\mu$.

Therefore, the analysis is now narrowed down to the determination of $\overline{T}_{trF_i}$, a parameter that, according to 4.1, is comprised of the fragment average collision time, $\overline{T}_C$, and the channel average busy time, $\overline{T}_{bu}$.

#### 4.1.3.1 Determination of $\overline{T}_C$

As illustrated in Fig. 4.2, $\overline{T}_{trF_i}$ is composed of three different durations: (i) the time interval from the beginning of the fragment transmission till the arrival of a PU on the channel: $t_C$, (ii) the interval from the PU's arrival till the next sensing instant: $t_W$, and (iii) the minimum interval since the beginning of the sensing instant that SU requires to gather enough samples for decision making regarding the presence/absence of PU: $t_D$. Therefore, it takes $(\bar{t}_W + t_D)$ for the SU to find out about the business of the channel and stop its transmission. This time could also be seen as the PU's interference duration $T_{int}$, in which the SU causes interference to the PU:

$$T_{int} = \bar{t}_W + t_D. \tag{4.9}$$

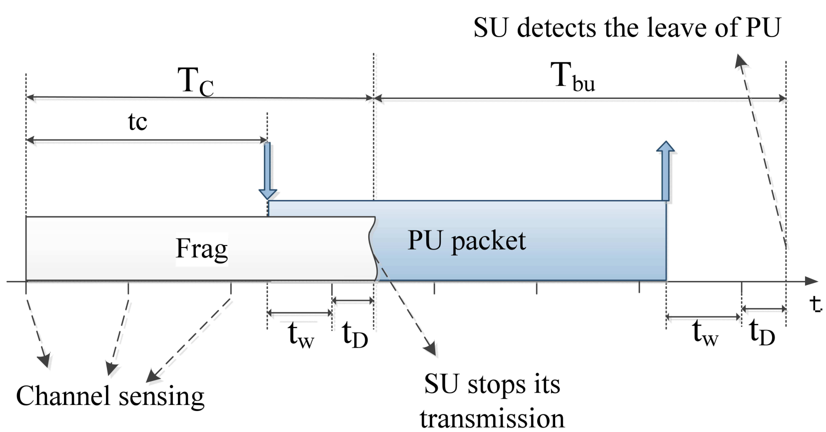



Figure 4.2: A collision situation.

With regard to the first term of $\overline{T}_{trF_i}$, i.e., $t_C$, the cumulative distribution function (CDF) of $t_C$ can be calculated based on (4.4) and the memoryless property of the exponential distribution as follows:

$$P\left(t_C \leq y \,|\, 0 \leq t_C \leq \frac{L_f}{R}\right) = \frac{P\left(0 \leq t_C \leq y\right)}{P\left(0 \leq t_C \leq \frac{L_f}{R}\right)} = \frac{1 - e^{-\alpha y}}{1 - e^{-\alpha L_f/R}}. \tag{4.10}$$

We note that the condition in (4.10) simply reflects the fact that the PU's arrival must occur somewhere during $L_f/R$ so that $t_c < L_f/R$, otherwise it is meaningless to define $t_C$. Next, the probability density function (PDF) of (4.10) is:

$$f\left(y\right) = \frac{\alpha e^{\alpha y}}{1 - e^{\alpha L_f/R}}. \tag{4.11}$$

Consequently, $\bar{t}_C$ is given by

$$\bar{t}_C = E\left(t_C\right) = \int_0^{L_f/R} y f(y) dy = \frac{1}{\alpha} - \frac{\left(L_f/R\right) e^{-\alpha L_f/R}}{1 - e^{-\alpha L_f/R}}. \tag{4.12}$$

By using the memoryless property of the exponential distribution once more, we can also define $t_W$ for the fixed interval $[0,\, \delta]$ according to

$$P\left(t_W \geq y \,|\, 0 \leq t_W \leq \delta\right) = \frac{P\left(y \leq t_W \leq \delta\right)}{P\left(0 \leq t_W \leq \delta\right)}, \tag{4.13}$$

and the mean of (4.13) could be derived as

$$\bar{t}_W = \delta - \frac{1}{\alpha} + \frac{\delta e^{-\alpha\delta}}{1 - e^{-\alpha\delta}}. \tag{4.14}$$

Finally, to quantify $t_D$, we note that $t_D$ is related to the minimum number of samples, $S_{min}$, that the secondary terminal must pick up from the received signal.

Representing the correct detection probability with $P_d$ and the false alarm probability with $P_f$, then $S_{min}$ is obtained using (4.15), [93]:[2]

$$S_{min} = \frac{1}{\gamma^2}\left(Q^{-1}\left(P_f\right) - Q^{-1}(P_d)\right)\sqrt{2\gamma+1}, \tag{4.15}$$

where $Q$ is the $Q$-function defined as $Q\left(x\right) = \int_x^\infty e^{-\frac{x^2}{2}}dx$ and $\gamma$ is the SNR of the received signal. As a result, $t_D$ is found as $t_D = S_{min}/F_s$, with $F_s$ denoting the sampling frequency.

Putting all these terms together, we can now find the total fragment collision interval as

$$\overline{T}_C = \overline{t}_C + \overline{t}_W + t_D. \tag{4.16}$$

#### 4.1.3.2 Determination of $\overline{T}_{bu}$

$T_{bu}$ is simply defined as the time that SU finds the channel occupied by PU. By referring to Fig. 4.2 once more, it is clear that this quantity is equal to $1/\mu$ (time between the arrival and departure of the PU) with only a shift of $\overline{t}_W + t_D$. So in general, we can consider $\overline{T}_{bu}$ as

$$\overline{T}_{bu} = \frac{1}{\mu}. \tag{4.17}$$

By having all the necessary variables, we can now find $\overline{T}_{trF_i}$. The point of departure is as follows: the transmission of a fragment is successful in the first trial with the probability $p_t/1-(1-p_t)^m$ for having a length of $L_f/R$, or in the second trial with probability $p_t(1-p_t)/1-(1-p_t)^m$ for having a length of $L_f/R+\overline{T}_{bu}+\overline{T}_c$, and so forth till the $m^{\text{th}}$ trial with probability $p_t(1-p_t)^{m-1}/1-(1-p_t)^m$ for a length of $L_f/R+(m-1)\left(\overline{T}_C+\overline{T}_{bu}\right)$. Thus, the average successful fragment transmission length is given by

$$\overline{T}_{trF_i} = \sum_{j=1}^{m}\frac{p_t(1-p_t)^{j-1}}{1-(1-p_t)^m}\left(\frac{L_f}{R}+(j-1)\left(\overline{T}_C+\overline{T}_{bu}\right)\right). \tag{4.18}$$

According to (4.18), $\overline{T}_{trF_i}$ is not a function of $i$, so we can simply call it $\overline{T}_{trF}$ and, finally, the total successful packet transmission time would be:

$$\overline{T}_{tr} = N\overline{T}_{trF}. \tag{4.19}$$

[2]This equation is derived for the PSK modulation.

### 4.1.4 Packet Dropping Time

As discussed earlier, a packet drop occurs if any fragment of that packet is dropped. The procedure for calculating the average drop time, $\overline{T}_d$, is similar to the one used before. Here, if the packet drop is due to the dropping of fragment number one, then $\overline{T}_d$ will be equal to the average drop time of fragment number one. Otherwise, if the dropping is due to fragment number two, $\overline{T}_d$ would be the sum of the average time of successful transmission for fragment number one and the mean drop time of fragment number two, and so forth. Thus, averaging over the all possible cases, we get

$$\overline{T}_d = p_t^N N\overline{T}_{trF} + \sum_{j=0}^{N-1} p_t^j \left(1-p_t\right)\left((m-1)\overline{T}_{bu} + m\overline{T}_C + j\overline{T}_{trF}\right). \tag{4.20}$$

### 4.1.5 Average Successfully Transmitted Data Fraction

In this section, we aim at finding the fraction of data packet that gets successfully transmitted before the presence of a PU causes interruption of the secondary's transmission. This quantity, denoted $F_T$, will help us compare the FFD and FD networks against each other. As the starting point, we note that the statistical independence among different fragments of a packet implies that the number of successfully delivered fragments ($N_T$) until the first interruption is distributed according to a truncated geometrical distribution with parameter $p_t$:

$$f\left(N_T = i\right) = \begin{cases} p_t^i(1-p_t) & 0 < i < N \\ p_t^N & i = N. \end{cases} \tag{4.21}$$

On the other hand, since $F_T = N_T/N$, then it is easy to calculate the mean of this quantity using (4.22), bearing in mind that with the probability $p_t^i(1-p_t)$ only $i/N$ of the packet get delivered successfully.

$$\overline{F}_T = \sum_{i=0}^{N-1} p_t^i(1-p_t)\frac{i}{N} + p_t^N = \frac{p_t - p_t^{N+1}}{N(1-p_t)}. \tag{4.22}$$

### 4.1.6 Energy Efficiency

Calculating the amount of consumed energy in successful or unsuccessful transmission states is directly related to the amount of time that the system spends in each of these states. Specifically, if a packet transmission requires a power of $P_{tr}$ and the sensing requires a power of $P_{sens}$, then it is not difficult to calculate the consumed energy during the periods $L_f/R$, $\overline{T}_c$ and $\overline{T}_{bu}$, respectively, as shown below.

$$E_f = \frac{L_f}{R} P_{tr} + \left\lfloor \frac{L_f}{R\delta} \right\rfloor t_D P_{sens}. \tag{4.23}$$

$$\overline{E}_C = \overline{T}_C P_{tr} + \left\lfloor \frac{\overline{T}_C}{\delta} \right\rfloor t_D P_{sens}. \tag{4.24}$$

$$\overline{E}_{bu} = \left\lfloor \frac{\overline{T}_{bu}}{\delta} \right\rfloor t_D P_{sens}. \tag{4.25}$$

One should recall that in FD mode and during the times $L_f/R$ and $\overline{T}_c$, the transmitter and the receiver are both active, one for the sake of transmitting and the other for the purpose of sensing the channel. However, during $\overline{T}_{bu}$, only the sensing task is to be performed so that the departure of PU can be detected.

Using the energy terms in (4.23)-(4.25) and substituting the relevant intervals (4.16)-(4.18), the total energy consumption for a successful fragment transmission $\overline{E}_{trF}$, the total energy consumption for successful packet transmission $\overline{E}_P$, and the average wasted energy because of the packet drop $\overline{E}_d$ are obtained as follows:

$$\overline{E}_{trF} = \sum_{i=1}^{m} \frac{p_t \left(1-p_t\right)^{i-1}}{1-\left(1-p_t\right)^m} \left(E_f + (i-1)\left(\overline{E}_C + \overline{E}_{bu}\right)\right), \tag{4.26}$$

$$\overline{E}_P = p_t^N N \overline{E}_{trF}, \tag{4.27}$$

$$\overline{E_d} = \sum_{i=1}^{N-1} p_t (1-p_t)^{i-1} \left((m-1)\,\overline{E}_{bu} + m\overline{E}_C + i\overline{E}_{trF}\right). \tag{4.28}$$

Finally, the quantity of interest, i.e., the system energy efficiency factor, is calculated as

$$\rho = \frac{E_{useful}}{E_{total}} = \frac{\overline{E}_P}{\overline{E}_d + \overline{E}_P}. \tag{4.29}$$

## 4.2 Simulation Results and Model Validation

In this section, we compare the performance of cognitive operation using the proposed FFD scheme, with the FD and HD schemes, based on the derivations developed in this chapter. The following table shows the parameter setting of the simulations.

Table 4.1: Simulation Parameters.

| Parameter | FFD | FD & HD |
|---|---|---|
| $\mu$ | 110 $s^{-1}$ | 110 $s^{-1}$ |
| $L_p$ | 2000 & 7000 Bytes | 2000 & 7000 Bytes |
| $L_o$ | 550 Bits | – |
| $L_f$ | 400 Bytes | – |
| $m$ | 3 | 3 |
| $\gamma$ | 0.031 | 0.031 |
| $P_d$ | 0.9 | 0.9 |
| $P_f$ | 0.05 | 0.05 |
| $P_{tr}$ | 110 mW | 110 mW |
| $P_{sens}$ | 80 mW | 80 mW |
| $\delta$ | 0.1 ms | 0.1 ms (FD), 100 ms (HD) |

The PU's arrival rate is chosen as the abscissa in all the subsequent figures due to the importance of PU activity level in the analysis.

As stated at the beginning of the paper, the main goal of utilizing FD over HD in CRNs is to reduce the interference onto PUs. Fig. 4.3 clearly proves the inferiority of using HD in CR context, in terms of interference time. For the HD scheme, this time can be derived from (4.9) and (4.14) with a given value for sensing interval time.[3] As reported in Fig. 4.3, the interference time in the HD-based scheme is enormously higher (scale is logarithmic) than its FD counterpart.

Fig. 4.4 is a comparison of the fraction of the effective transferred data $\overline{F}_T$, which has been derived in subsection 4.1.5, before the first PU arrival in FFD vs. FD schemes, taking the PU arrival rate as parameter. As observed, the results of both schemes are the same for low PU arrival rates. However, as $\alpha$ increases, $\overline{F}_T$ drops sharply for the FD scheme compared to the FFD scheme implementing fragmentation. This is due to the fact that an FD secondary user in a busy network would not get a chance to transmit its data all at once and the whole packet would be dropped in the presence of PU on the channel. As such, the average successful data fraction that could be transmitted in such scheme would decrease abruptly.

[3]In the provided figures, plots pertaining to the HD scheme are obtained using a sensing interval time of 100 $ms$. This time can vary depending on different channel characteristics and system parameters, but we have checked the results for a wide range of values ($1-1000$ $ms$) and confirmed the same trends as observed for the plots shown here.

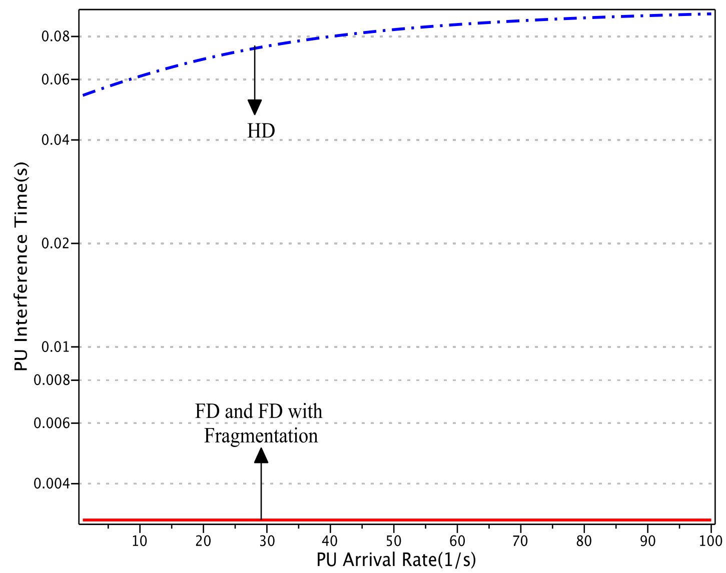



Figure 4.3: PU interference time, $T_{int}$, in HD and FD systems.

When $\overline{F}_T$ decreases, the average packet dropping rate could increase in each packet transmission. Therefore, this figure leaves no doubt that the FFD scheme is superior, particularly for a highly active network of PUs.

For a better understanding of the performance of the FFD scheme, we compare the average successful packet transmission time $\overline{T}_{tr}$ (derived in subsection 4.1.3) for all three methods in Fig. 4.5. Apparently, the time of successful transmission for a FFD network is slightly larger than in the FD system; however it has a huge difference with HD especially for higher $\alpha$. One should not forget that although this figure reports a better performance for the FD scheme, from Fig. 4.4 we already know that its packet dropping rate is way worse than that of the scheme implementing fragmentation.

Finally, the energy efficiency of the three schemes is illustrated in Fig. 4.6. Here, we use two different data lengths to show the impact of the packet length on performance. Clearly, there is not much of difference for the energy efficiency of FFD for a data length of 2000 and 7000 bytes and the two corresponding plots overlap. However, the plots of FD and HD have sudden changes and drop abruptly by increasing the packet length. For a data length of 7000 bytes, we observe that the HD scheme has almost the same efficiency as the FD system and it is also reasonably close to FFD, especially for low and high values of $\alpha$. Hence, in general, it seems that the FFD system works better than the FD with no fragmentation in a network with high traffic and for longer data packets. Having almost the same performance as FD and lower packet dropping rate, and also providing less interference to PU, we conclude

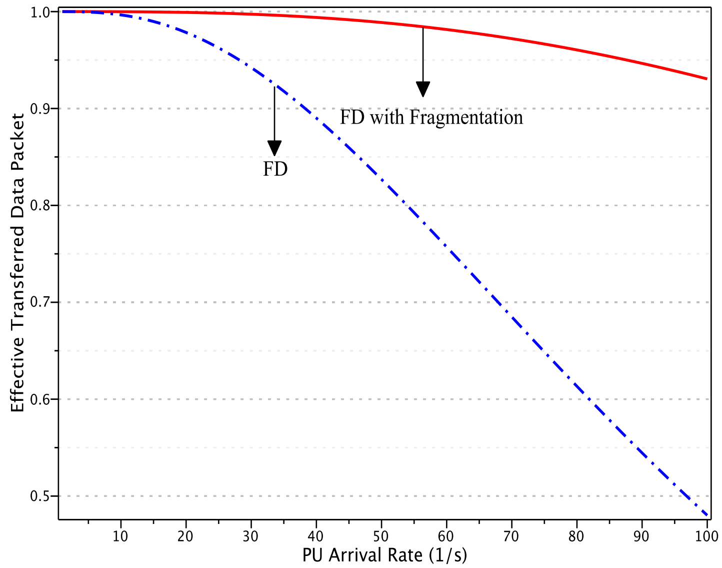



Figure 4.4: Average data fraction that is successfully transmitted over the channel before the first PU arrival, $\overline{F}_T$.

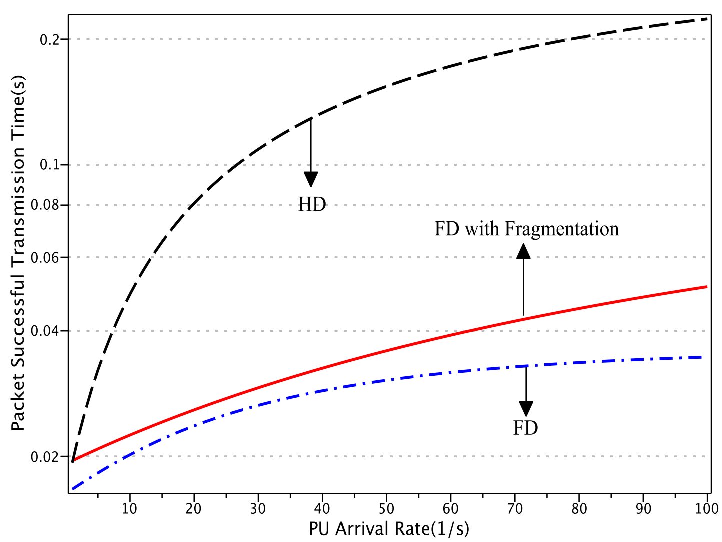



Figure 4.5: Packet successful transmission time $\overline{T}_{tr}$: HD vs. FD with or without fragmentation.

that the fragmentation-enabled FD scheme, i.e., FFD, is a good choice for the next generation of CR systems.

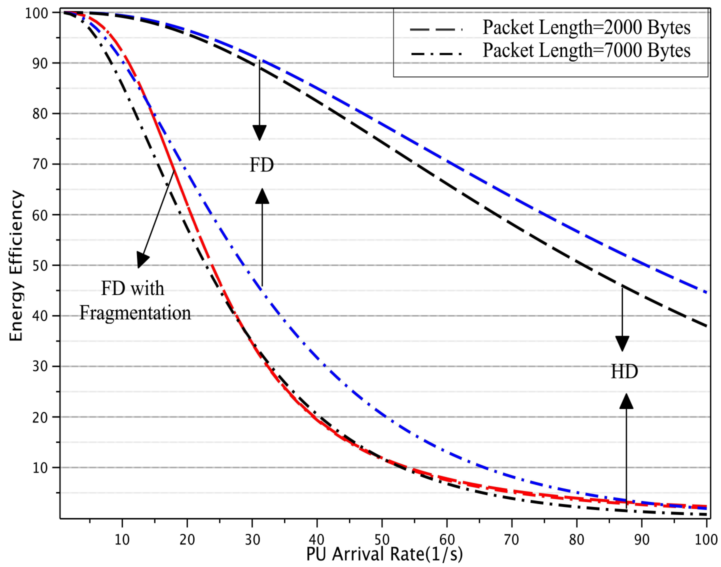



Figure 4.6: Energy efficiency $\rho$ for data lengths of 2000 and 7000 bytes: HD vs. FD with or without fragmentation.

# CHAPTER 5

# DFD-MAC: PROTOCOL DESCRIPTION AND PERFORMANCE ANALYSIS

In this chapter, we propose a decentralized access MAC protocol called distributed-access full-duplex MAC (DFD-MAC) that leverages the full advantages of FD. At the same time, our proposed MAC can address the limitations of the CONTRA MAC protocol [4] such as the high probability of collision and the limitation on the number of the hops that the data packet can traverse after winning the channel ( Section 3.2).

To reduce the hidden terminal effects, the data header can be sent prior to the rest of the packet along the path in order to guarantee the reservation of the path before the data transmission starts. This technique also known as the worm-hole routing technique [9], [28] is going to be used in the proposed MAC in conjunction with an un-contended access mechanism [4]. Moreover, the DFD-MAC protocol is capable of forwarding the packets in a route for an arbitrary number of hops, unlike the approach in [4] which is confined to two-hop configurations only. This is an efficient approach, designed to take full advantage of the opportunities that the single-channel FD in the PHY layer opens up for having true spectral efficiency.

In the first part of this chapter, we introduce the proposed protocol, explain the way it works and describe its advantages over other existing protocols. Then in section 5.2, we model our protocol using a discrete-time Markov chain (DTMC) similar to Bianchi modeling. Section 5.3 is devoted to the mathematical derivations to find the path throughput and end-to-end delay in the network and finally, we validate our scheme by comparing the forenamed metrics with HD CSMA/CA counterparts in section 5.3. Numerical results are provided and show that our protocol achieves better performance than CSMA/CA operating in HD mode.

## 5.1 Protocol Description

As mentioned before, the access mechanism of DFD-MAC is based on CSMA/CA. The reason for this choice is the efficiency, flexibility and scalability that CSMA/CA equips the network with. Having that in mind, here in the DFD-MAC, once the wireless node finishes its back-off countdown and wins the contention, it forwards a control packet, called a forwarding route request (FRR) packet, to its destination in a similar way as the RTS is transmitted in CSMA/CA mechanism. This FRR packet has the same length ($T_{FRR}$) as the RTS and carries the same information such as source MAC address, destination MAC address and NAV. Further, we assume the route is already determined by the network layer. Thus, in addition to all information each FRR carries, it also contains the MAC address of the next node through which the data is going to traverse. Finally, the FRR packets help to shrink the collision window to $T_{FRR}$ and, hence, increase the probability of successful transmission.

The post-contention transmission in DFD-MAC is decomposed into two phases: (i) path establishment by sequential FRR control packets and (ii) data transmission.

### 5.1.1 Path Establishment

As shown in Fig. 5.1, during the first phase, after node $A$ wins the contention, it starts transmitting the FRR packet to node $B$. Once $B$ hears FRR-$A$, it starts transmitting FRR-$B$ to node $C$, which also counts as an implicit ACK to node $A$, helping it to imply that its packet was already successfully received by $B$. This backward overhearing is important in the sense that it enables the hop-head node $A$ to transmit its data to the hop-end node $B$, during the data transmission phase, only if this implicit ACK is received by $A$ (cf. Fig. 5.1). The FRR forwarding operation in the first phase is carried out by decoding FRR-$A$, stripping off the control packet, extracting the information inside of it, replacing the source and destination MAC addresses and NAV properly, encoding the packet again (now FRR-$B$ is generated), and forwarding the packet to the next hop-end which is $C$. Afterwards, the forwarding process for the following hops continues accordingly, as shown in Fig. 5.1. It is to be noted that according to our MAC protocol, wherever a hop-head node (say $B$) along the path cannot overhear the transmission of FRR (say FRR-$C$) from its immediate upstream hop-end node (say $C$), the hop-head node assumes that its FRR control packet (say FRR-$B$) was not successfully decoded by its hop-end due to collision or hidden terminal effect. Therefore, the path and un-contended data delivery can only extend up to that point (say $B$), requiring it to participate in the next contention.

### 5.1.2 Data Transmission

The second phase in the DFD MAC commences while the first phase still continues. In fact, as shown in Fig. 5.1, once the FRR handshaking is still going on two hops farther from the path initiator ($A$) down the way, data transmission from $A$ to $B$ starts with the rationale that the ongoing transmissions of FRRs are far enough not to interfere with the aforementioned data transmission at the location of the receiver $B$. When $B$ receives the data packet, it waits until $T_Y/2$ seconds elapse from the beginning of the transmission from $A$, where $T_Y$ is the time that it takes for the data packet with length $Y$ to be transmitted . The role of FD technology is going to be highlighted here. In fact, it empowers $B$ to transmit at the same time as $A$ does, but with a minimum time shifting $T_Y/2$. The latter is required to avoid interference due to the concurrent transmissions from $A$ and $C$, which prevents $B$ from being able to decode the rest of $A$'s transmissions. By applying this suitable time shifting in the data forwarding procedure, we are able to extend the reach of un-contended communications beyond two hops.[1]

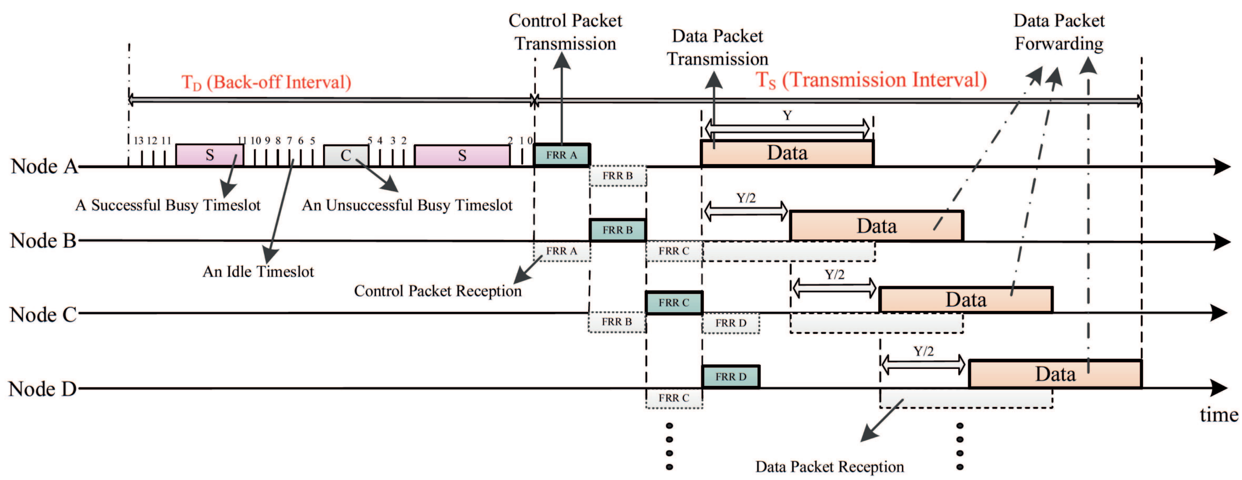



Figure 5.1: The protocol timing and FD operation when a data packet progresses in four hops contending for channel access only once.

### 5.1.3 Protocol Features

As it is implied, the central advantages of our proposed protocol are threefold: *(i)* it spends less resource on transmitting control packets (implicit ACK); *(ii)* it opposes the hidden terminal problem by firstly establishing the route, which results in shrinking the collision window's length; and *(iii)* it easily adapts itself to changing

[1]We note that in the protocol, there is no need of using an ACK to inform the transmitter about the successful receive of data. According to Fig. 5.1, node $A$ can make sure about the successful transmission of data to node $B$ only by listening to node $B$'s transmission. Hence, if $A$ hears the end of node $B$'s packet forwarded to $C$, it means that the whole packet has already been received successfully by $B$.

situations while maximizing the progress. As an illustration of the last property, when the number of stations is larger, and therefore collision occurrence is high, the protocol autonomously prevents the progress through many hops since somewhere along the path, the high chance of collision cuts the route to a path with lower hop-count.

In addition to the previous advantages, the DFD-MAC protocol has the ability of path refining as well, which means that if the routing layer makes a mistake in defining the optimized route, the protocol has the ability to correct it. As shown in Fig. 5.2, let us say that nodes $B$ and $C$ are both in the hearing range of node $A$, the path provided from the routing layer is $\boldsymbol{A \rightarrow B \rightarrow C}$ though the data can be sent from $A$ directly to $C$. After sending the FRR-$A$, if node $A$ hears both FRR-$B$ and FRR-$C$, it concludes that node $C$ is also in its hearing range and, thus, can decide to change the IP of the node in the data packet and send the data straight to node $C$. Therefore, the MAC layer is able to optimize the path that is offered by the routing layer.

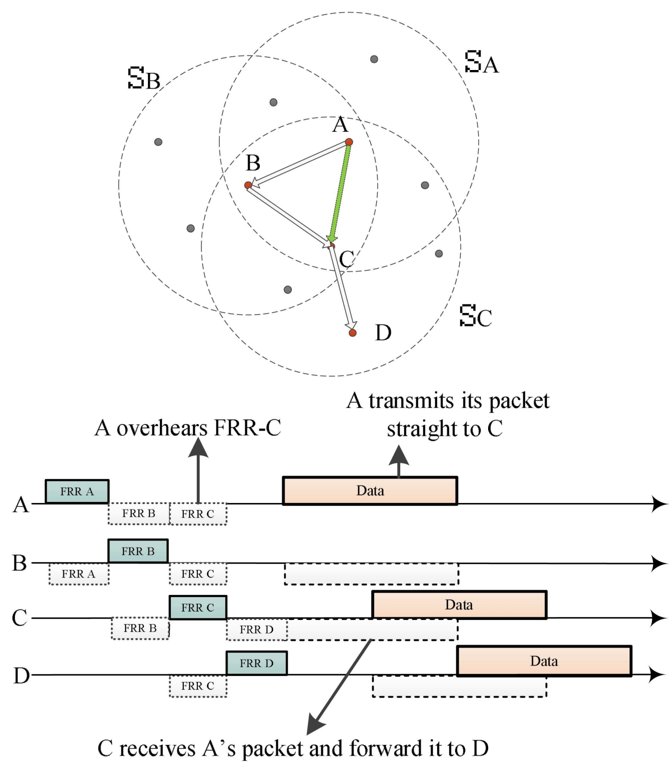



Figure 5.2: The ability of the proposed protocol to refine the route assigned by the network layer.

Finally, we should mention that DFD-MAC can be easily converted to the regular CSMA/CA at any time that the FD technology faces a problem and the necessity of switching to HD mode is inevitable. Thus, toggling a bit in the FRR packet can define the HD or FD mode of the transmitter and for the case of HD mode, another bit would represent if the control packet is an RTS or CTS. The rest of transmission can then be done in CSMA/CA protocol format.

## 5.2 Protocol Modeling

To characterize the dynamics of the proposed protocol and analyze its performance, we need to establish a model based on the Markov chain, which is discrete in time and state (DTMC). The DTMC depicted in Fig. 5.3 characterizes the behavior of a single node in the network running the DFD-MAC protocol described above. Taking advantage of the aforementioned resemblances between the access mechanism of our MAC protocol and CSMA/CA, the DTMC in this thesis is built upon the model introduced in [94] and [7]. According to the theory of Markov chains, this DTMC has a steady state solution since it is non-null recurrent and its number of states is finite.

According to CSMA/CA, before sending data on the channel, each node should choose a random back-off number from the contention window depending on the transmission history. This window is defined as $(0, W_i)$ where $W_i = 2^i W_0$, $i = 1, 2, \ldots, m$, and $W_0$ is the minimum value which is used for the first stage of transmission when the packet has not experienced any unsuccessful transmission prior to that. Depending on the stage of retransmission (say $i$), the contention window expands up to the maximum value of $2^m W_0$, which is called the maximum back-off window. At this instant, if a packet faces more unsuccessful transmissions, the contention window does not expand anymore. We assume no retransmission limit, meaning that a collided packet will be retransmitted until it is successfully received.

Given this brief overview, in our DTMC, each state is represented by a triplet $(i, k, l)$ where $i$ is the number of retransmission stages that the node experiences and $k \in (0, W_i)$ represents the back-off count. According to the DFD-MAC protocol description, the transmissions states in this DTMC are divided into two types; the node is either transmitting its own data if it is the winner of contention (like node $A$ in Fig. 5.1) or forwarding another node's packet (like nodes $C$, $D$,... are doing in Fig. 5.1). These state types are correspondingly represented in dark-blue and gray colors in Fig. 5.3. With this knowledge, $l$ characterizes these two categories whereby $l = 0$ indicates that the node is transmitting its own data packets while $l = 1$ indicates that the node is in forwarding mode. Therefore, $P_{i,k,l}$ represents the probability of a single state of this DTMC.

To explain the interactions in the DTMC, let us start from a node that is currently in the state $(0, W_0 - 1, 0)$. This node may stay in this state when the channel is busy, with the channel business probability $P_{bu}$, or it may go to state $(0, W_0 - 1, 1)$ if it is decided that it must participate in forwarding another node's data packet, with probability $P_f$. If none of these events happens, the node decrements its back-off and moves to state $(0, W_0 - 2, 0)$ with probability $1 - P_c$, which indicates that the channel is sensed idle. Of course, since there is no other possibility, the probabilities' sum of the transitions going out of a state should be one. Assuming the node already forced to do the forwarding task (according to the protocol description presented before) and entered $(0, W_0 - 1, 1)$, then the node stays in this state with probability $q$ and leaves it when the forwarding task is over, with probability $1 - q$.

Once the back-off counter reaches zero, the node is allowed to transmit its own data packet, which goes through with probability $1 - p$ and fails with conditional collision probability $p$. Given the above explanations, after writing the state equations, the closed-form solution for $P_{i,k,l}$ is derived as follows:

$$P_{i,k,l} = \frac{W_i - k}{W_i} \left(\frac{P_f}{1-q}\right)^l \times \begin{cases} \dfrac{1-p}{1-P_c} \displaystyle\sum_{j=0}^{m} P_{j,0,0} & i = 0 \\ \left(\dfrac{p}{1-P_c}\right)^i P_{0,0,0} & 1 \leq i \leq m-1 \\ \dfrac{p}{1-P_c} \left(P_{m-1,0,0} + P_{m,0,0}\right) & i = m \end{cases} \tag{5.1}$$

where

$$P_{i,0,0} = \left(\frac{p}{1-P_c}\right)^i P_{0,0,0} \quad , 0 \leq i \leq m-1 \ , \tag{5.2}$$

$$P_{m,0,0} = \left(\frac{p}{1-P_c}\right)^m \frac{1-P_c}{1-P_c-p} P_{0,0,0}. \tag{5.3}$$

Defining $\tau_1$ as the total transmission probability (the sum of probabilities associated with the dark-blue states in Fig. 5.3) of a node for dispatching its own data, we obtain

$$\tau_1 = \sum_{i=0}^{m} P_{i,0,0} = \frac{1-P_c}{1-P_c-p} P_{0,0,0} \left(1 - \left(\frac{p}{1-P_c}\right)^{m-1} + \left(\frac{p}{1-P_c}\right)^m\right). \tag{5.4}$$

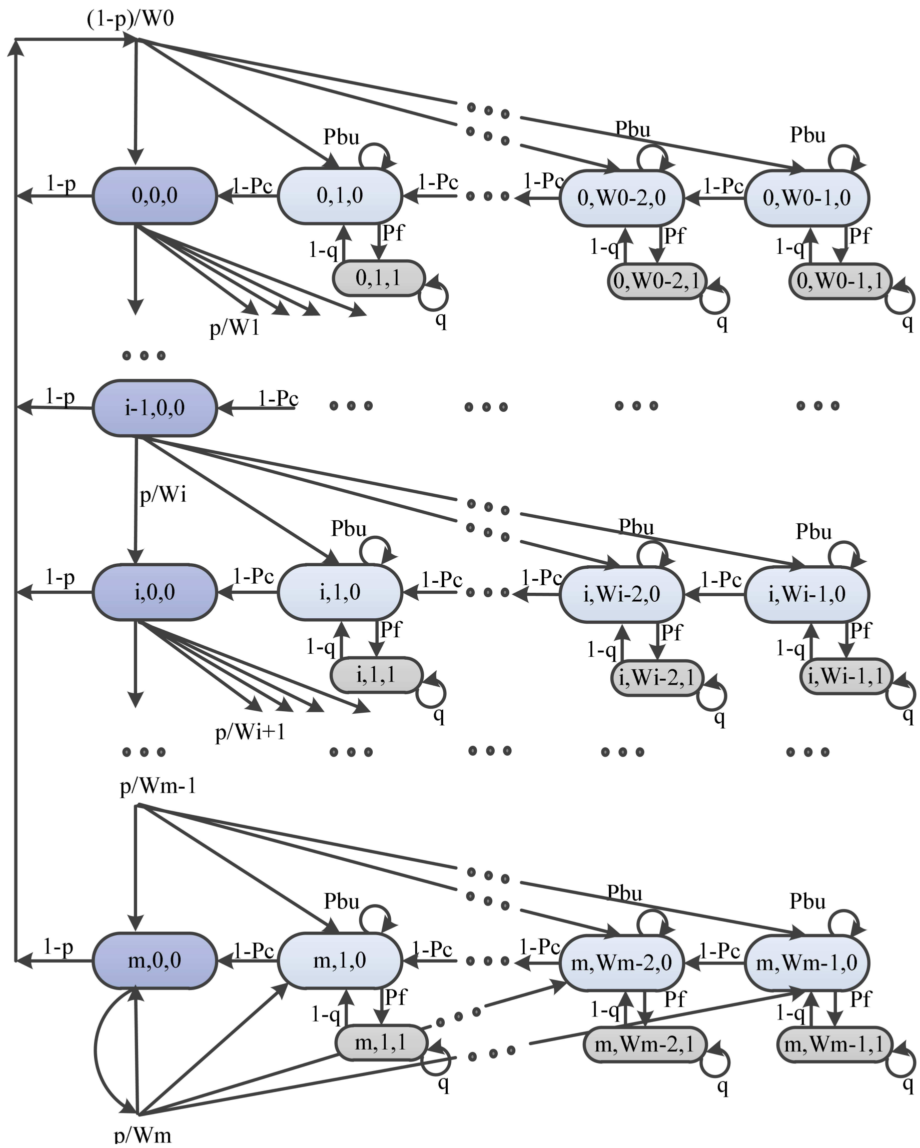



Figure 5.3: The proposed DTMC characterizing the dynamics of the DFD-MAC protocol at each node.

Also, let $\tau_2$ represent the forwarding probability, which is equal to the sum of probabilities pertaining to the gray states (cf. Fig. 5.3):

$$\tau_2 = \sum_{i=0}^{m} \sum_{k=1}^{W_i-1} P_{i,k,1}. \tag{5.5}$$

Therefore, the total transmission probability $\tau = \tau_1 + \tau_2$ would be as shown in (5.6).

$$\begin{aligned}\tau = &\left(1-\frac{p}{1-P_c}\right)^{-1} P_{0,0,0} + \frac{P_f}{1-q} P_{0,0,0} \left(\left(1-\frac{p}{1-P_c}\right)^{-1} \left(\frac{1-p}{1-P_c} \cdot \frac{W_0-1}{2}\right)\right) + \\ &\frac{P_f}{1-q} P_{0,0,0} \left( \frac{(1-P_c)\,W_0}{2} \left( \frac{\frac{2p}{1-P_c} - \left(\frac{2p}{1-P_c}\right)^m}{1-\frac{2p}{1-P_c}} - \frac{\frac{p}{1-P_c} - \left(\frac{p}{1-P_c}\right)^m}{1-\frac{p}{1-P_c}} \right)\right) + \\ &\frac{P_f}{1-q} P_{0,0,0} \left( \frac{1-P_c}{1-P_c-p} \cdot \frac{2^m W_0 - 1}{2} \right).\end{aligned} \tag{5.6}$$

Now in order to find the value of $\tau$, it is necessary to determine the value of $P_{0,0,0}$. This is accomplished using the normalization condition

$$\sum_{L=0}^{1} \sum_{i=0}^{m} \sum_{k=0}^{W_i-1} P_{i,k,l} = 1\,, \tag{5.7}$$

where after substituting $P_{i,k,l}$ from (5.1) into (5.7) and using (5.2)-(5.4), we obtain (5.8).

$$\begin{aligned}&(P_{0,0,0})^{-1} = \\ &\frac{1-P_c}{1-P_c-p} \left( \left(1-\left(\frac{p}{1-P_c}\right)^{m-1} + \left(\frac{p}{1-P_c}\right)^m\right) + \left(1+\frac{P_f}{1-q}\right)\frac{1}{2}\left(\frac{p}{1-P_c}\right)^m (2^m W_0 - 1)\right) \\ &+ \frac{1-P_c}{1-P_c-p}\left(\left(1+\frac{P_f}{1-q}\right)\left(\frac{1-p}{1-P_c}\right)\left(\frac{W_0-1}{2}\right)\left(1-\left(\frac{p}{1-P_c}\right)^{m-1} + \left(\frac{p}{1-P_c}\right)^m\right)\right) + \\ &\frac{1-P_c}{2(1-P_c-p)}\left(1+\frac{P_f}{1-q}\right)\left(\frac{W_0\,(1-P_c-p)\left(\frac{2p}{1-P_c}\right) - \left(\frac{2p}{1-P_c}\right)^m}{1-P_c-2p} - \frac{p}{1-P_c} + \left(\frac{p}{1-P_c}\right)^m\right).\end{aligned} \tag{5.8}$$

### 5.2.1 Derivation of DTMC's Unknowns

Before getting into the derivation of the remaining unknowns in (5.8), we should first give a brief explanation on the topology of the network under consideration and then determine the number of contending stations in the network. To do so, we assume that nodes are deployed according to 2D Poisson distribution with density $\lambda$ (nodes/m$^2$). Therefore, supposing that $S_A$ is the area of node $A$'s hearing range (with radius $R$), then the number of nodes that is positioned in $S_A$ (including $A$ itself) is given by

$$N_A = \lambda S, \tag{5.9}$$

where $S = \pi R^2$. Since transmission ranges of all nodes are assumed to be the same (meaning that nodes transmit with the same power), then according to (5.9), the number of neighbours would be the same for all nodes, as is for node $B$. To find how many nodes are located in the intersection of the hearing ranges of nodes $A$ and $B$ (denoted $N_{A\cap B}$), we need to know the area of this intersection ($S_{A\cap B}$). By geometry, given that the transmission range is $R$ and that two nodes are located $a$ units apart from each other (with $2R > a$ held for two nodes to hear each other), this quantity is obtained as,

$$S_{A\cap B} = 2\left(cos^{-1}\left(\frac{a}{2R}\right) - \frac{a}{2R}\sqrt{1-\left(\frac{a}{2R}\right)^2}\right). \tag{5.10}$$

Since, as a fact, in a Poisson point process inside a circle, the distance from the center to an arbitrary chosen point (distanced at $a$) inside that circle has the following probability density function (PDF)

$$f_a\left(a\right) = \frac{2a}{R^2} \quad 0 < a < R, \tag{5.11}$$

then, using (5.11) to remove the condition in (5.10) and having in mind that $N_{A\cap B} = \lambda\, S_{A\cap B}$, we get

$$N_{A\cap B} = \left(\pi - \frac{3\sqrt{3}}{4}\right)\lambda R^2. \tag{5.12}$$

Finally, $N_{B-A\cap B}$, which is the number of nodes in the area $S_B - S_A$, would be equal to $N_A - N_{A\cap B}$.

Now that we determined the number of contending nodes, we can characterize the unknown probabilities in (5.8). Starting with the collision probability $p$, it can be calculated as

$$p = 1 - \left(1-\tau\right)^{N_B}\left(1-\tau\right)^{(U-1)N_{B-A\cap B}}, \tag{5.13}$$

where $U = T_{FRR}/\delta$ is the number of time slots that the transmission of the FRR packet lasts and $\delta$ is the slot length. The reason (5.13) is derived that way is explained as follows: when node $A$ transmits the FRR packet to node $B$ (Fig. 5.1), all nodes in $S_B$ may cause collision (total of $N_B$) by transmitting over the channel in the first slot of the FRR. However, only those located in $S_B - S_A$ (total of $N_{B-A\cap B}$) can cause a collision during all the remaining slots $(U-1)$ of the FRR since they weren't able to hear FRR-$A$ so that they stop counting down the back-off and, thus, they are considered hidden terminals to $B$. A similar approach has been taken in [95] to analyse the CSMA/CA protocol with the existence of the hidden terminal. Given that, the channel business probability is given by

$$P_{bu} = \left(1 - (1-\tau)^{N_A}\right) \frac{N_A - 1}{N_A}, \tag{5.14}$$

which means that the transmission from at least one node in $S_A$ (excluding a forwarding request assigned to $A$) keeps the channel busy.

A generic node goes into a forwarding state if there is one transmission in its corresponding hearing range. Suppose that one node can only forward a packet once, during the time that it wins the contention to transmit its own data packet; we can define $P_f$ as

$$P_f = \frac{\tau(1-p)}{\overline{BO}}, \tag{5.15}$$

where $\overline{BO}$ is the average number of idle back-off slots [96], which according to the DTMC in Fig. 5.3 would be equal to

$$\begin{aligned}\overline{BO} = &(1-p)\, W_0 + p\,(1-p)\,(W_0 + W_1) + \ldots + p\,(1-p)^m\,(W_0 + W_1 + \ldots + W_m) + \\ &\sum_{v=1}^{\infty} p\,(1-p)^{m+v}\,(W_0 + \ldots + W_m + v\, W_m),\end{aligned} \tag{5.16}$$

with $W_i = 2^i W_0$ for $i = 1, 2\ldots, m$. Then after simplification, (5.16) reduces to:

$$\overline{BO} = \frac{W_0\,(1-p)}{2p}\left(\frac{1-2p^{m+1}}{1-2p} - \frac{1-p^m}{1-p} + p^{m+1}\left(\frac{2^m-1}{1-p} + \frac{2^{m-1}}{1-p} + p\frac{2^{m-1}}{(1-p)^2}\right)\right). \tag{5.17}$$

Having found $P_f$ and $P_{bu}$, it is easy to determine $1-P_c$ since $P_f+P_{bu}+1-P_c=1$. As it is obvious, up to now the DTMC probabilities depend on $\tau$ and this makes the equations for finding $\tau$ recursive.

Finally, to fully characterize (5.8), it is necessary to find $q$, which is the probability that a node stays in forwarding mode (gray states in Fig. 5.3). The average number of times that the node returns to the forwarding state is equal to the mean of a geometric probability mass function (PMF) and is equal to $1/(1-q)$. On the other hand, this average is to be equal to the number of slots for having a successful transmission in the forwarding mode (with average length $\overline{T}_{s,Fw}$, to be derived later), which is equal to $\overline{T}_{s,Fw}/\delta$. Equating these two quantities to find $q$, we obtain,

$$q = \frac{\overline{T}_{s,Fw} - \delta}{\overline{T}_{s,Fw}}. \tag{5.18}$$

As described in the previous subsection, when a node goes into forwarding mode, its transmission could be successful or may encounter a collision. Thinking a bit deeper, we can easily find out that the length of this successful forwarding depends on the position of the forwarding node along the path. For example, according to Fig. 5.1, the length $T_{s,Fw}$ for node $B$ is equal to $2T_{FRR}+3Y/2$ and it is different from $T_{s,Fw}$ pertaining to node $C$, which is equal to $T_{FRR}+2Y$, with $Y$ standing for the data packet length. Consequently, for the $n^{th}$ hop of the path, we can write

$$T_{s,Fw_i} = (3-n)\,T_{FRR} + \frac{n+2}{2}Y. \tag{5.19}$$

Therefore, depending on the location of the forwarder with respect to the chosen path, different forwarding lengths are experienced. Since the number of hops ($1 < n < N$) that a data packet can traverse is a random variable with PDF

$$P_{Fw}(n) = \frac{p\,(1-p)^{n-1}}{1-(1-p)^N}, \quad n = 1, \ldots, N, \tag{5.20}$$

where $N$ is the maximum number of hops along a path that a packet is allowed to traverse, then the average number of hops that a data packet can proceed ($\bar{n} < N$) without needing to re-contend is given by

$$\bar{n} = \frac{1-(1-p)^N\,(1+Np)}{p(1-(1-p)^N)}. \tag{5.21}$$

Now using (5.20) in order to average over all the possibilities in (7.22), we get $\overline{T}_{s,Fw}$ as follows:

$$\overline{T}_{s,Fw} = \sum_{n=2}^{N} \left( (4-n)\, T_{FRR} + \frac{n+1}{2} Y \right) \left( \frac{p\,(1-p)^{n-1}}{1-(1-p)^{N}} \right). \tag{5.22}$$

Equations (5.9) to (5.22) provide all the unknown quantities required to characterize $P_{0,0,0}$ in (5.8). Thus, after plugging (5.8) into (5.6), a recursive expression relating the transmission probability $\tau$ to all the network parameters, $N$, $W_0$, $m$, etc., is obtained. Solving this equation by numerical methods results the value of $\tau$, thus completing the analytical derivations of our DTMC model.

## 5.3 DFD-MAC Performance Analysis

### 5.3.1 Performance Metrics

We start the performance evaluation of the proposed protocol by defining pertinent quantities such as throughput and delay. The prevalent single-hop definition of throughput is not applicable to this case anymore and a multi-hop definition should be proposed. Therefore, we define the normalized path throughput $TH_{path}$ as the effective data payload that can be transmitted along a path during a period of time (denoted $T_t$ for now), as follows:

$$TH_{path} = \frac{E\,(Y)}{T_t}. \tag{5.23}$$

There are different alternatives for the selection of the appropriate values for $T_t$ and $E\,(Y)$, where the choice of one always determines the value of the other. For example, in many works, such as [7], the denominator is fixed, $T_t = \delta$, and throughout is defined as the effective payload transmitted during a tiny time slot.

The approach that we choose in this thesis is to find the delay that a data packet experiences while it is being delivered from source to destination along the forwarding path. During this time interval, also called path delay ($D_{path}$), a fixed amount of information is definitely delivered ($E\,(Y) = c$) and, as such, can be considered as an alternative definition for throughput. Hence, letting $T_t = D_{Path}$, where

$$D_{Path} = T_D + T_S, \tag{5.24}$$

it is possible to derive the throughput of the system with $T_S$ being the average successful transmission time of a data packet along the path and $T_D$ the contention interval that node $A$ has to wait till it can capture the channel and starts transmission.

It should be noted that throughput is always a quantity that can be defined for the owner of traffic, which is node $A$ (the path head), and that is why it is incorrect to apply the classical definition of throughput here. In other words, the metric of throughput cannot be defined for the forwarder nodes (such as $B$, $C$,...) over the traffic that they do not own but are only involved in forwarding.

As shown in Fig. 5.1, based on the approach in [96] the contention time $T_D$ is composed of four intervals. These components are: (i) $A$'s wasted collision time $T_C$, during which the previous transmission trials from $A$ have not been successful; (ii) the neighbours' total successful time $T_{S'}$, during which $A$ should freeze its timer since the channel is busy with successful transmission from others; (iii) the neighbours' total collision time $T_{C'}$, during which $A$ freezes its timer because of channel business with unsuccessful transmission from others; and (iv) idle time $T_I$, during which the channel is not busy. As a result, $T_D = T_C + T_{S'} + T_{C'} + T_I$. In the next part, we are going to derive each of these intervals gradually for the scenario portrayed in Fig. 5.1.

### 5.3.2 Interval Derivations

#### 5.3.2.1 $\mathbf{T_S}$

This is the total time spent for successful transmission starting from node $A$. Evidently, $T_S$ depends on the number of hops that the data can move along Fig. 5.1. Given that the average number of hops that the data packet can traverse is $\bar{n}$, then $T_S$ can be calculated as follows:

$$T_S = \left(3T_{FRR} + 2Y + \frac{\bar{n}-2}{2}Y\right). \tag{5.25}$$

#### 5.3.2.2 $\mathbf{T_C}$

This is the total time wasted in unsuccessful transmissions from $A$. Such transmission events occur when $A$ does not hear the node $B$'s FRR that is destined for $C$. Define $\overline{n}_c$ as the average number of times that $A$'s packet faces collision and $\bar{t}_c$ the duration of each collision, then the total time of unsuccessful transmissions is

$T_C = \overline{n}_c \cdot \overline{t}_c$. Since the number of trial times that node $A$ is allowed to retransmit a packet is geometrically distributed, according to the DTMC, then

$$P(i) = (1-p)\, p^i \quad , i = 0,\, 1, \ldots \tag{5.26}$$

with the average $\overline{n}_c = p/(1-p)$. To find $\overline{t}_c$, it is sufficient to take a close look at Fig. 5.1. As shown, if node $A$'s FRR packet is collided at $B$, after the time of 2 $T_{FRR}$ elapses and if $A$ does not hear FRR-$B$, it realizes that FRR-$A$ has not been decoded by $B$ in the last hop and that is why $B$ did not forward FRR-$B$. Hence, $A$ makes sure that it should re-contend with a new back-off counter to get another transmission chance. This being said,

$$T_C = \frac{p}{1-p}\, 2\, T_{FRR}. \tag{5.27}$$

#### 5.3.2.3 $\mathbf{T_{S'}}$

This is the total time spent in successful transmissions from neighbours of $A$. Defining $\overline{n}_{s'}$ as the average number of these transmissions, and $\overline{t}_{s'}$ as the average length of each transmission, then $T_{S'} = \overline{n}_{s'} \cdot \overline{t}_{s'}$. Supposing that there are $k_1$ successful transmissions and $k_2$ unsuccessful transmissions out of $j$ busy slots (during the contention period $T_D$), then $\overline{n}_{s'}$ can be found according to Binomial distribution (because of independence) of having $k_1$ successful transmissions out of $j$ busy slots, as

$$\Pr\left(n_{s'} = k_1 | j\right) = \begin{pmatrix} j \\ k_1 \end{pmatrix} \left(1 - q_{c'}\right)^{k_1} \left(q_{c'}\right)^{j-k_1}, \tag{5.28}$$

where $q_{c'}$ is the probability of unsuccessful transmission when the header node (say $A$) senses the channel busy by the packet forwarding of a path in its vicinity (say $A' \to B' \to \cdots$). Hence, $q_{c'}$ can be derived as

$$q_{c'} = \frac{P_{bu} - (1-\tau)^{N_{B'}}\, (1-\tau)^{(U-1)N_{B'-A'\cap B'}}}{P_{bu}}, \tag{5.29}$$

with $P_{bu}$ given in (5.14). Here, $N_{B'}$ and $N_{B'-A'\cap B'}$ stand for the number of nodes in $S_{B'}$ and $S_{B'-A'\cap B'}$, respectively, similar to the definitions in (5.9) and (5.12). Of course, $q_{c'}$ would not change if node $A$ has been exposed to hear the other parts of the path, because of the primary assumption of having the same hearing range for all the nodes.

The conditional PDF in (5.28) has the average $E\left(\overline{n}_{s'} = k_1 | j\right) = (1 - q_{c'})\, j$. Since the probability of having $j$ busy slots in $BO$ total time slots is again binomially distributed due to the independence assumption, i.e.,

$$\Pr\left(j \text{ busy slots} \,|\, BO = b\right) = \binom{b}{j} P_{bu}^{j} (1 - P_{bu})^{b-j}, \tag{5.30}$$

then, after the probabilistic simplifications, we obtain

$$\begin{aligned}\overline{n}_{s'} &= E_{BO}\left(E_j\left(E\left(n_{s'} = k_1 | j\right)\right)\right) \\ &= \sum_{BO=0}^{\infty} \sum_{j=0}^{BO} \{E\left(n_{s'} = k_1 | j\right) \Pr\left(j \text{ busy slots} \,|\, BO = b\right) \Pr\left(BO = b\right)\} = (1 - q_{c'}) P_{bu} \overline{BO}.\end{aligned} \tag{5.31}$$

Finding $\bar{t}_{s'}$, as needed for $T_{S'} = \overline{n}_{s'} \cdot \bar{t}_{s'}$, is not a straightforward task since, for a given full-duplexing path, node $A$ (as a listener) might be exposed to transmissions from forwarders located in a different fraction of that path. Fig. 5.4 illustrates all different possibilities for a three-hop path. For example, in Fig. 5.4a, $A$ is exposed to node $A'$ transmission for the length of $t_{s'a}$, while in Fig. 5.4b, node $A$ hears node $B'$'s transmission for the duration of $t_{s'b}$. Both of these times are illustrated in Fig. 5.4g with node $A$ freezing its counter for the length of one transmission. For

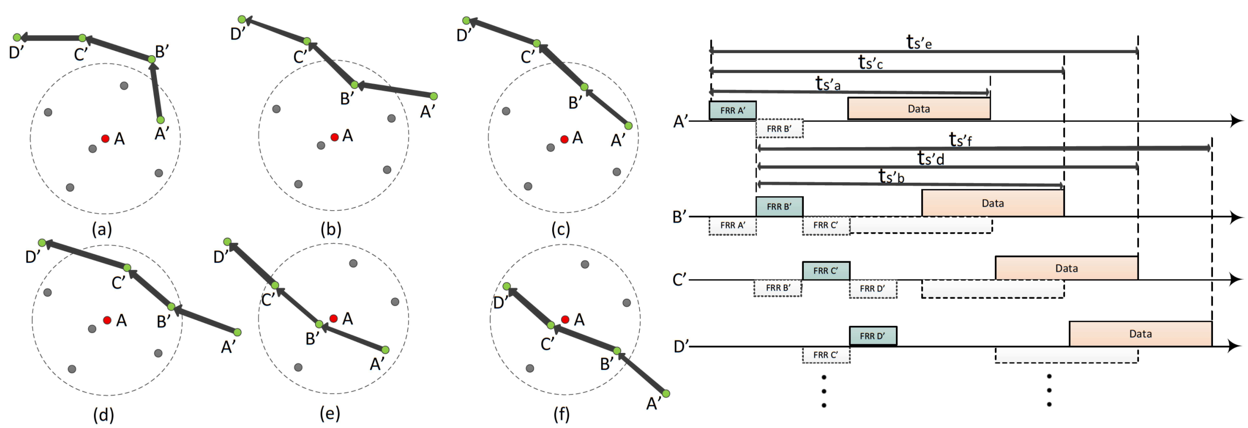



Figure 5.4: Different placement possibilities of a hearer node ($A$) w.r.t. a forwarding path.

the case in Fig. 5.4c, $A$ hears transmissions from two nodes (the first and second of the path) and, thus, what is heard is from the start of node $A^{'}$'s transmission till the end of node $B^{'}$'s forwarding, which is equal to $t_{s'c}$ (Fig. 5.4g). For sure, $\bar{t}_{s'}$ would be different in Fig. 5.4d, where node $A$ can hear node $B^{'}$'s and $C^{'}$'s transmissions. Consequently, the length of $\bar{t}_{s'}$ (overheard by $A$) depends not only on the number of nodes in its neighbourhood (area $S_A$), but also on the position of the nodes. Given this description, the time of successful transmission $t_{s'z}(z = 1, 2, 3)$ heard by $A$ when one, two, or three forwarders of this path are located inside $S_A$ is equal to:

$$\begin{cases} t_{s'1} = (4-n)\,T_{FRR} + \dfrac{n+1}{2}Y & N \geq 1 \\ t_{s'2} = (5-n)\,T_{FRR} + \dfrac{n+1}{2}Y & N \geq 2 \\ t_{s'3} = (6-n)\,T_{FRR} + \dfrac{n+1}{2}Y & N \geq 3 \end{cases} \tag{5.32}$$

where $n$ is the highest-order node in the path whose transmission can be heard by node $A$ (e.g., $n = 2$ in Fig. 5.4c and $n = 4$ for the case in Fig. 5.4f) and $N$ is the maximum number of hops along a path that a packet is allowed to traverse. It should be noted that cases (a) and (b) in Fig. 5.4 can occur for any choice of $N \geq 1$, that (c) and (d) can occur when $N \geq 2$, and that (e) and (f) can occur when $N \geq 3$, hence the way (5.32) is written.

Now by averaging (5.32) using weighting probabilities (5.20), the average of successful transmission times $\bar{t}_{s'z}(z = 1, 2, 3)$ for each category is obtained as follows:

$$\begin{cases} \bar{t}_{s'1} = \sum\limits_{n=1}^{N} \dfrac{p\,(1-p)^{n-1}}{1-(1-p)^{N}} \left((4-n)\,T_{FRR} + \dfrac{n+1}{2}Y\right) \\ \bar{t}_{s'2} = \sum\limits_{n=2}^{N} \dfrac{p\,(1-p)^{n-1}}{1-(1-p)^{N}} \left((5-n)\,T_{FRR} + \dfrac{n+1}{2}Y\right) \\ \bar{t}_{s'3} = \sum\limits_{n=3}^{N} \dfrac{p\,(1-p)^{n-1}}{1-(1-p)^{N}} \left((6-n)\,T_{FRR} + \dfrac{n+1}{2}Y\right) \end{cases} \tag{5.33}$$

However, in order to find the total $\bar{t}_{s'}$, we still need to know the probability of node $A$ being exposed to one $(\Pr(z = 1))$, two $(\Pr(z = 2))$ or three $(\Pr(z = 3))$ nodes of a neighbour path.

Next, we introduce a simplified solution to calculate these unknowns, using the geometry theory. Suppose that there is a source, a destination, and a route linking them up, and assume the hearer node $A$ is being exposed to part of this route, as depicted in Fig. 5.5. By the assumption that hearing ranges are the same and nodes that create a path do not deviate much (not more than $\varepsilon$ as shown in Fig. 5.5) from the connection line (dashed line in the figure), it is clear that there are at most three nodes of a path located inside $S_A$. This assumption is not unrealistic at all since it is based on the fact that the network layer acts reasonably and never determines a route that makes small progresses in each hop or wastes resources by revolving around a node.

For the case of $z = 1$, it is necessary for the connection line to fall in the confined area, between the circle and cord of the length $R$ on the top and bottom of the circle (Fig. 5.5). If the connection falls inside the tiny middle strip with width of $\varepsilon$, the case $z = 3$ is realized with three nodes inside $S_A$. For the remaining area of the circle, the case $z = 2$ is realized with two nodes inside $S_A$.

Consequently, the probability of having $z$ nodes ($z = 1, 2, 3$) of a neighbour path located in the hearing range of node $A$ is equal to the dashed area (for each case) divided by the total area of $\mathrm{S_A}$. Therefore,

$$\Pr(z=1) = \frac{2\left({}^{1}\!/_{6} \cdot \pi - {}^{\sqrt{3}}\!/_{4}\right) R^2}{\pi R^2}, \tag{5.34}$$

$$\Pr(z=3) = \frac{2\varepsilon \cdot 2R}{\pi R^2}, \tag{5.35}$$

$$\Pr(z=2) = 1 - \Pr(z=1) - \Pr(z=3). \tag{5.36}$$

Ultimately, we can calculate the final value of $\bar{t}_{s'}$ by averaging (5.33) using (5.34)-(5.36) as follows:

$$\bar{t}_{s'} = \sum_{z=1}^{3} \bar{t}_{s'z} \Pr(z). \tag{5.37}$$

#### 5.3.2.4 $\mathbf{T_{C'}}$

This quantity corresponds to the total time spent in unsuccessful transmission from neighbours of $A$. As $\overline{n}_{c'}$ is the average number of unsuccessful transmissions of the neighbours, and $\bar{t}_{c'}$ is the average length of each transmission, then $T_{C'} = \overline{n}_{c'}\bar{t}_{c'}$.

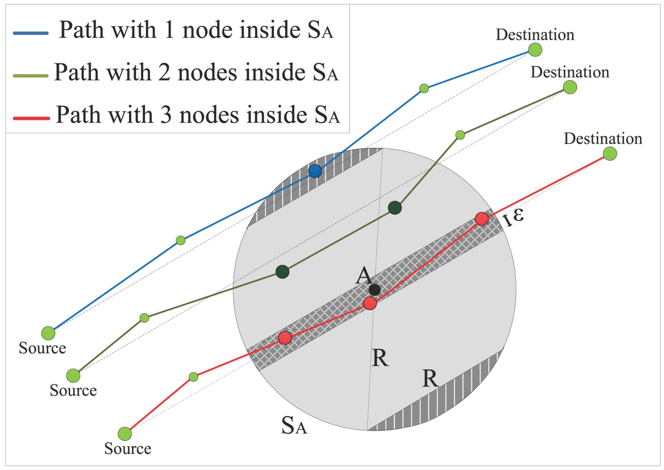



Figure 5.5: Node $A$ different placement to a path and hearing one, two or three nodes of that path.

To find $\overline{n}_{c'}$, suppose that there are $k_2$ unsuccessful transmissions out of $j$ busy slots. Then according to the binomial distribution, we have

$$\Pr\left(n_{c'} = k_2 | j\right) = \binom{j}{k_2} \left(q_{c'}\right)^{k_2} \left(1 - q_{c'}\right)^{j-k_2}, \tag{5.38}$$

with $q_{c'}$ given in (5.29). Similar to what we did in (5.28)-(5.31), the average $E\left(n_{c'} = k_2 | j\right) = q_{c'} \cdot j$. Using (5.30), the probability of having $j$ busy slots during $BO$ slots, we have

$$\begin{aligned} \overline{n}_{c'} &= E_{BO}\left(E_j\left(E\left(n_{c'} = k_2 | j\right)\right)\right) \\ &= \sum_{BO=0}^{\infty} \sum_{j=0}^{BO} \{E\left(n_{c'} = k_2 | j\right) \Pr\left(j \text{ busy slots} \mid BO = b\right) \Pr\left(BO = b\right)\} = q_{c'} P_{bu} \overline{BO}. \end{aligned} \tag{5.39}$$

Similar to $\bar{t}_{s'}$, $\bar{t}_{c'}$ consists of three terms ($\bar{t}_{c'z}$ , $z = 1, 2, 3$), which are the neighbours' average collision times in case of one, two or three node(s) in the hearing range of $A$. Thus, to find $\bar{t}_{c'1}$, using (5.20) and similar to (5.33), we write:

$$\bar{t}_{c'1} = \sum_{n=1}^{N} \frac{p(1-p)^{n-1}}{1-(1-p)^N} \left( (4-n)\, T_{FRR} + \frac{n-1}{2} Y \right) \quad N \geq 1. \tag{5.40}$$

The case of two nodes in $S_A$ is divided into two subcategories; the packet might collide either in the first or second hop. For example, in Fig. 5.4c the packet might collide in $A'$ and never be received by $B'$. The other possibility is that the packet is successfully received by $B'$ but collided during its transmission to node $C'$. Therefore, both cases should be considered in the calculation of $\bar{t}_{c'2}$:

$$\begin{aligned}\bar{t}_{c'2} =& \sum_{n=1}^{N-1} \frac{p(1-p)^{n-1}}{1-(1-p)^N} \left((4-n)\,T_{FRR} + \frac{n-1}{2}Y\right) \\ &+ \sum_{n=2}^{N} \frac{p(1-p)^{n-1}}{1-(1-p)^N} \left((5-n)\,T_{FRR} + \frac{n-1}{2}Y\right) \qquad N \geq 2.\end{aligned} \tag{5.41}$$

A similar explanation is also valid for three nodes in $S_A$. According to Fig. 5.4e, the collision might happen in $B'$, $C'$ or $D'$, and $t_{c'3}$ would be different for each case. Thus,

$$\begin{aligned}\bar{t}_{c'3} =& \sum_{n=1}^{N-2} \frac{p(1-p)^{n-1}}{1-(1-p)^N} \left((4-n)\,T_{FRR} + \frac{n-1}{2}Y\right) \\ &+ \sum_{n=2}^{N-1} \frac{p(1-p)^{n-1}}{1-(1-p)^N} \left((5-n)\,T_{FRR} + \frac{n-1}{2}Y\right) \\ &+ \sum_{n=3}^{N} \frac{p(1-p)^{n-1}}{1-(1-p)^N} \left((6-n)\,T_{FRR} + \frac{n-1}{2}Y\right) \qquad N \geq 3.\end{aligned} \tag{5.42}$$

Finally, similar to (5.37), to get the value of $\bar{t}_{c'}$ we use

$$\bar{t}_{c'} = \sum_{z=1}^{3} \bar{t}_{c'z} \cdot \Pr(z). \tag{5.43}$$

#### 5.3.2.5 $\mathbf{T_I}$

$T_I$ is the total time during which node $A$ finds the channel idle and counts down its back-off. This is equal to the average number of back-off slots $\overline{BO}$ in (5.17) multiplied by the slot length, i.e.,

$$T_I = \delta \cdot \overline{BO}. \tag{5.44}$$

All the above equations enable us to calculate the path throughput $TH_{path}$ of the network according to (5.23) which is a function of $N$. To be able to validate the performance of our proposed MAC protocol, we obtained the throughput of the 802.11 CSMA/CA network whose basic functionality (including freezing, back-off, etc.) is the same as our proposed MAC protocol in general. By letting the same parameters run for both, for fair comparisons, we solved the aforementioned 802.11 DTMC and found the path throughput and delay according to [94], [7] and [97], and also based on the definition in (5.23) after entailing the hidden terminal problem as stated in [95] and [98]. In fact, the hidden terminal is the dominant source of errors in multi-hop ad-hoc networks and should not be neglected at all.

## 5.4 Model Validation

To validate the efficiency of our protocol and get a better understanding of its performance, the throughput of the proposed MAC protocol will be compared with CSMA/CA. The parameter settings are according to Table I, and as we mentioned before, values of common parameters are the same.

Table 5.1: Simulation Setup

| **Parameter** | **CSMA/CA** | **DFD-MAC** |
|---|---|---|
| $W_0$ | 64 | 64 |
| $m$ | 8 | 8 |
| Packet Payload | 1024 bytes | 1024 bytes |
| PHY header | 128 bits | 128 bits |
| RTS/FRR | 160 bits | 160 bits |
| CTS | 112 bits | – |
| ACK | 112 bits | – |
| CTS Timeout | 300 $\mu$s | – |
| ACK Timeout | 300 $\mu$s | – |
| SIFS | 28 $\mu$s | – |
| DIFS | 128 $\mu$s | – |
| Slot Time | 50 $\mu$s | 50 $\mu$s |
| Data Rate | 11 Mb/s | 11 Mb/s |

As the equations state, the path throughput and delay of the DFD-MAC is a direct function of $N$, which is the number of hops destined for a path. Hence, the transmission might be cut off in the way to the final destination because of the collision and hidden terminal effects. To that end, we build the results based on the average number of hops ($\bar{n}$) that the packet could traverse along the path.

In Fig. 5.6 and Fig. 5.7, we compare the path delay and throughput results for different numbers of neighbouring nodes (shown as $N_N$ in the figures). According

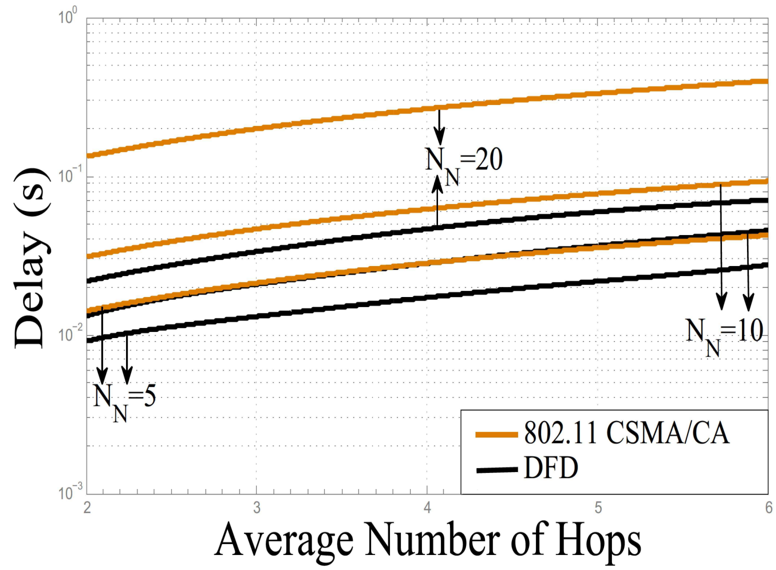



Figure 5.6: Path delay of DFD-MAC and 802.11 CSMA/CA for different numbers of neighbour nodes.

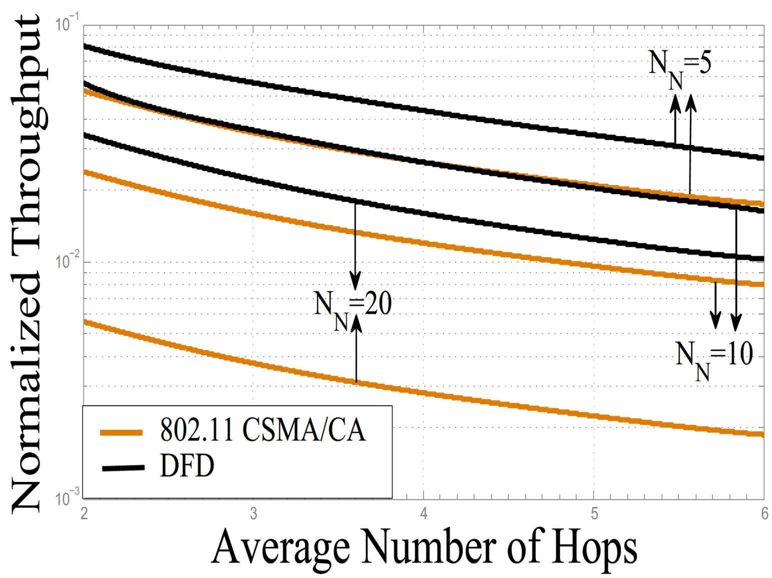



Figure 5.7: Path throughput of DFD-MAC and 802.11 CSMA/CA for different numbers of neighbour nodes.

to Fig. 5.6, as the number of neighbour nodes decreases, the path delay of both the CSMA/CA protocol and our FD MAC decreases. Of course, this trend is expected since the probability of collision always increases for a denser network (higher number of neighbouring nodes), and as the result of that, the packet faces more delay while being forwarded. However, we can see that the performance of our MAC protocol is better than its CSMA/CA in each case, with a difference in performance that becomes larger as the average number of hops increases. The reason can be nothing but the overlaps between the transmission of different hops and the chance of having un-contended transmission for the forwarding nodes. The same line of reasoning can be given for Fig. 5.7 to justify the performance improvement of the DFD-MAC, which is two times better on average.

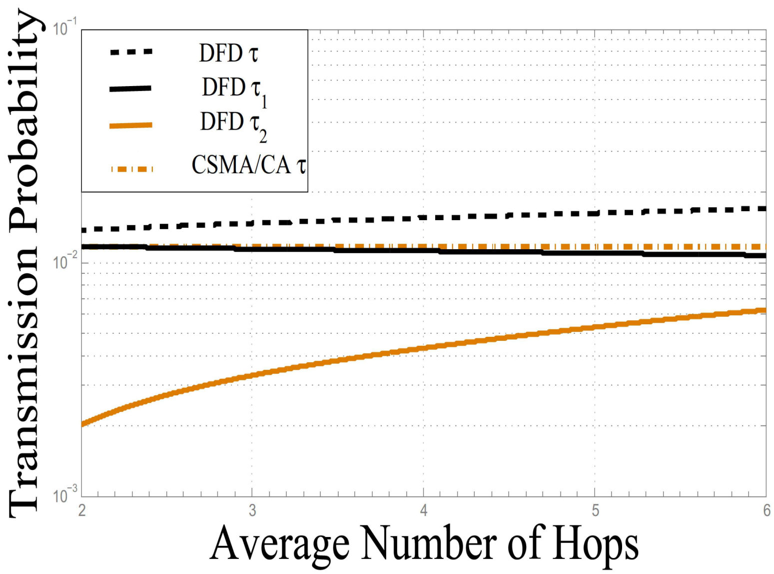



Figure 5.8: Comparison of DFD-MAC transmission probabilities ($\tau_1$, $\tau_2$, $\tau$) with the transmission probability of CSMA/CA.

Fig. 5.8 reports the transmission probability extracted from the DTMC while the number of neighbouring nodes equals 10 for both CSMA/CA and DFD-MAC. As observed, the transmission probability of a node sending its own packet in DFD-MAC ($\tau_1$), is almost equal to the transmission probability of CSMA/CA. In other words, the chance of a node to start a path in DFD-MAC is as with CSMA/CA. However, the total transmission probability ($\tau$) for DFD-MAC is much higher because of the additional term it comprises, namely, the transmission probability of the forwarding packets ($\tau_2$).

As we know, the ascending trend of $\tau$ in Fig. 5.8 has a negative impact on the performance of the network because of its main role in increasing the probability of packet collision and channel business, which may yield lower throughput for DFD-MAC protocol compared to CSMA/CA. However, this conclusion cannot be true as the ascending trend of $\tau_2$ helps improve the path throughput in a way that the chance of taking the forwarder role by a random node in the network increases. The total effect is the generation of longer paths that can forward the data packet in a shorter transmission time while also eliminating the contention time required in each hop to access the channel. This can compensate the increase of channel business and packet collision probability in the DFD-MAC and results in what Fig. 5.6 and Fig. 5.7 report.

Finally, by comparing both protocols, we can state that although the packet collision and channel business probability is higher in our protocol, its performance is still much better than the 802.11 and the only reason is the FD technology that makes the system able to bear the transmissions overlap of consecutive hops.

# CHAPTER 6

# CONCLUSION

In this thesis, we investigate the applications that full-duplex technology brings into the communication networks. While chapters 3 and 2 are introduction on full-duplex technology, CR networks and also Medium Access Layer, in chapter 4 we proposed a full-duplex fragmentation-enabled scheme targeting CR networks. The proposed cognitive operation exploits the FD advantages while reducing the packet drop rate (increases the amount of effective information transfer) triggered by fragmentation. Also, using the ability of transmitting and sensing the channel at the same time helps primary users to face a lower probability of collision for their data packets. The performance of this scheme was analyzed and compared against full-duplex (FD) and half-duplex (HD) schemes. In terms of the energy efficiency, the performance of this scheme is comparable to HD and FD without fragmentation for long data packet lengths. Hence, the use of the FD fragmentation-enabled scheme is highly appealing, especially for the scenarios with long data packets or in highly active primary networks.

In chapter 5, we presented a new MAC protocol for full-duplex distributed access networks. The protocol utilizes the advantages of full-duplex while lessening the hidden terminal problems in multi-hop transmissions.

In this chapter, we claimed that this protocol has the following advantages:

- It spends less resource on transmitting control packets (implicit ACK);
- It opposes the hidden terminal problem by firstly establishing the route, which results in shrinking the collision window length;
- It can be convertible to HD mode by only toggling a bit of a control packet;
- It easily adapts itself to changing situations while maximizing the progress.

By using a DTMC model similar to Bianchi's DTMC, we were able to evaluate the achievable path throughput and delay and compare the results to the equivalent model for the 802.11 CSMA/CA protocol. Results and comparisons showed the performance gains of the proposed full-duplex MAC protocol.

# CHAPTER 7

# RÉSUMÉ

## 7.1 Introduction

Avec le développement rapide des systèmes de communications sans fil et les demandes croissantes continues pour des applications large bande, l'allocation actuelle des bandes de fréquences n'est pas optimale. Tandis que la recherche actuelle montre qu'une grande fraction du spectre a été utilisée inefficacement, la radio cognitive (RC) [30] a offert la solution d'utiliser le spectre d'une façon efficace.

Une des restrictions dans l'opération cognitive est que les utilisateurs secondaires (USs) doivent éviter d'interférence avec les utilisateurs principaux (UPs) (ces derniers ont favorisé l'accès au canal) en fournissant un niveau acceptable de qualité de service pour les utilisateurs cognitifs.

L'imposition de ces exigences sur l'opération cognitive nécessite que les nœuds cognitifs fonctionnent dans une bande de fréquences tant qu'elle est vide et libèrent cette bande quand les UPs deviennent actif. Ceci implique directement que les USs écoutent le canal périodiquement pour détecter la présence/absence des UPs. Cette écoute exige des périodes de silence intermittentes à planifier, pendant lesquelles les USs devraient seulement écouter le canal pour des occasions potentielles de transmission.

C'est évident que plus l'intervalle séparant les écoutes (appelé aussi l'intervalle de repos) est grand, la chance d'avoir des interférences aux UPs est plus grand.

D'autre part, l'intervalle de repos plus court apporte plus de fiabilité, mais reduit l'utilisation de canal car les USs devraient rester silencieux pour une plus grande fraction de temps. Ceci est équivalent à la perte de plus grandes occasions de transmission. De telles pénalités sont associées à la nature inhérente de communication half-duplex (HD) qui limite les nœuds sans fil pour transmettre ou recevoir; il n'est pas possible d'exécuter ces opérations simultanément.

Avec le développement récent de la technologie Full-duplex (FD), il est devenu possible d'éliminer ces contraintes et c'est devenu faisable pour un dispositif sans fil d'écouter (recevoir) et transmettre simultanément (en annulant l'auto-interférence). Donc il n'est pas nècessaire de consacrer des intervalles d'écoutes séparés dans le dispositif FD sans fil. Ceci double presque l'utilisation du canal et le débit de transmission, mais augmente la complexité du système [9].

Très récemment, les aspects de réalisation de la technologie FD ont été explorés dans plusieurs directions; tous prouvent des améliorations considérables. [9] et [13] et le journal [2] qui a apparu ensuite, ils sont les études pionnières qui ont utilisé des méthodes d'annulation analogues novatrices (l'annulation d'antenne et la soustraction de signal) pour une réalisation pratique de FD.

Les résultats prouvent que la combinaison de ces techniques avec d'autres méthodes peut réduire la quantité d'auto-interférence par 60 dB. Ceci est une amélioration substantielle comparée aux méthodes connues précédemment d'annulation de bruit [20], [4], [14], et l'annulation dans domaine numérique [15], [21].

### 7.1.1 La Technologie Full-duplex et la Radio Cognitive

Le progresse mentionné dans la technologie Full-duplex a rapidement révélé le besoin d'examiner les opportunitès que les couches plus hautes du protocole peuvent soutenir les systèmes FD ([2, 4, 21]). Les chercheurs ont revisité les anciennes études présentées dans [23] et ont profité de FD pour améliorer la performance et la pénétration des réseaux de radio cognitive (RRCs). En effet, l'utilisation directe de technologie FD a l'intérêt majeur dans RRCs où un US équipé avec l'émetteur-récepteur FD peut écouter le canal dans des intervalles courts en les transmettant.

Dans le cas où l'US détecte la présence d'un UP, il libere le canal immédiatement. Car il n'affecte pas les transmissions d'UP comparées au scénario avec HD où l'intervalle de vulnérabilité peut être aussi grand que l'intervalle de repos.

Dans ce contexte, [1] a proposé un plan de FD pour RRCs qui emploie trois méthodes d'annulation d'interférence connues : (i) l'annulation avec antenne, (ii) l'annulation d'interférence du Rf et (iii) l'annulation d'interférence dans domaine numérique et a comparé la performance de FD et HD. Dans [25], les auteurs ont proposé un plan d'écoute pour RRCs non-encoché équipé de FD et ont évalué la performance de ce système en présence d'annulation auto-interférence non idéal.

Nous citons aussi [21], [26] et [27] comme des études se concentrant uniquement sur les aspects physiques de l'union FD et des technologies RC, comme resource allocation.

### 7.1.2 FD et la Couche Contrôle d'Accèss au Support (MAC)

La technologie FD a rapidement prolongé son application aux domaines différents de communication sans fil, comme les réseaux de maille intérieurs [14], la radio cognitive [25], [1] et le routage multi-trajets [24]. D'autre part, le progrès fondamental que cette technologie a apportée à la couche physique a causé directement des changements majors des couches supérieures de protocole, particulièrement les couches de routage et MAC. Basé sur notre connaissance, [2, 4, 3] sont les principaux articles qui ont proposé des nouveaux protocoles de MAC pour cette nouvelle technologie de couche physique.

Dans le protocole MAC de [2] qui est conçu pour des réseaux d'accès centralisés, deux nœuds transmettent leurs données simultanément, si un nœud n'a pas de paquet pour l'autre, il envoie un ton occupé sur le canal pour éliminer le problème de terminal caché. Le protocole MAC de [3] est pour des réseaux d'accès centralisés aussi et il est basé sur trois méthodes : (i) un back-off partagé, (ii) espionnage d'en-tête et (iii) résolution d'affirmation virtuelle.

Pour des réseaux distribués, [4] a proposé un protocole de MAC pour FD basé sur CSMA/CA. Au mieux de notre connaissance, ce protocole, nommé le " CONTRA FLOW ", est le seul protocole MAC distribué que l'on a proposé jusqu'ici. Dans ce protocole, quand le nœud $A$ capture le canal selon le protocol CSMA/CA, il commence à transmettre son paquet au nœud $B$ et attend le minuteur primaire (MP) pour expirer. Aussitôt que le nœud $B$ ouvre l'en-tête, il commence à expédier le paquet au $C$ par une transmission sans contention.

Si $A$ peut entendre le paquet du $B$ avant que le minuteur n'expire, il continue la transmission. Autrement, il arrête et exécute sa transmission dans une autre contention. Alors, le nœud $B$ transmet les données au $C$ avec un paquet d'acquittement (ACK) à l'$A$, et après que la transmission de données du nœud $B$ est complétée, le nœud $C$ envoie un ACK au $B$. L'idée de [4] pour la permission à un nœud d'expédier des données immédiatement et sans contention est nouvelle, mais le défi d'haute probabilité de collision. Car, la deuxième saut est sévèrement exposée aux transmissions de terminaux cachés.

Aussi, la couche de MAC correspondante n'est pas conçue pour permettre le progrès plus de deux chemins, parce que deux sauts ne sont pas nécessairement le nombre optimal d'étapes pour un progrès sans contention.

### 7.1.3 Packet Fragmentation de la Radio Cognitive Full-duplex

Dans cette partie, nous cherchons à évaluer le rendement des US dans un cadre cognitif lorsque les nœuds cognitifs sont activés avec la technologie FD. Nous comparons la performance d'un tel système avec le système HD et nous démontrons la supériorité du premier. Bien que l'exploitation de FD améliore l'efficacité de bande passante et permet les USs de découvrir les occasions de transmission plus rapidment, un support des couches plus hautes est nécessaire. À cette fin, nous suggérons la communication progressive en appliquant la fragmentation de paquets à la couche MAC. Particulièrement nous montrons qu'en divisant le paquet dans plus petits segments indépendants, la performance du système, ainsi que la probabilité d'une transmission réussie et la fiabilité de système est améliorée considérablement.

#### 7.1.3.1 Fragmentation

Jusqu'ici, plusieurs études ont démontré les avantages que la fragmentation peut provoquer dans un réseau local sans fil [91], [92]. Par exemple, les résultats de [91] montrent que la taille de fragment optimale qui minimise la consommation d'énergie de système HD avec le MAC CSMA/CA est 300 octets pour des longueurs de paquet de 1500 octets, avec 550 bits de "overhead" qui est exigé sur chaque fragment.

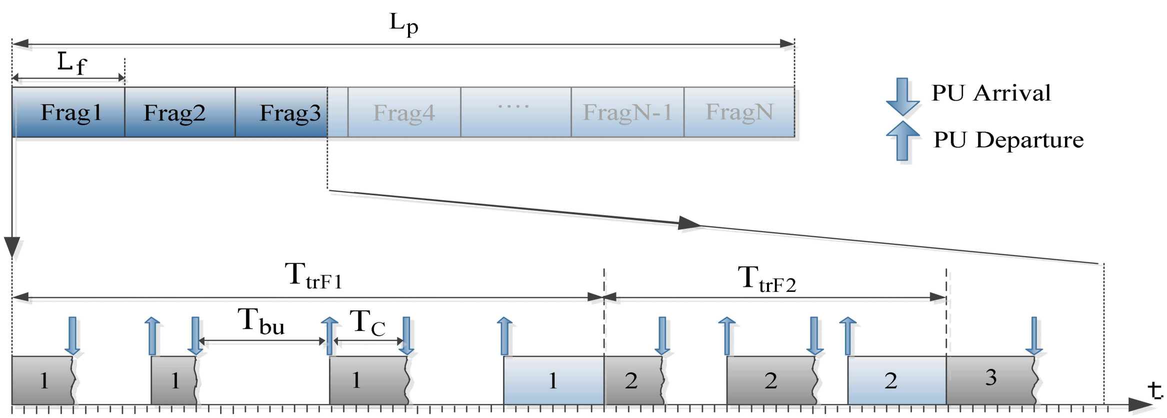



Figure 7.1: Fragmentation des paquets et sa transmission sur le canal.

Dans cette étude, nous définissons $L_f$ comme la longueur d'un fragment et $N$ comme le nombre de fragments dans un paquet. Alors, les fragments devraient être transmis dans l'ordre, cela signifie que la transmission du fragment suivant ne commencerait à moins que les fragments précédents soient déjà livrés. On permet à chaque fragment d'être retransmis jusqu'à $m$ fois si la présence des UPs cause l'avortement de la transmission initiale écarté et le paquet entier est baissé si un fragment ne peut être transmis avec succès après les $m$ fois de retransmission. Dans ce mémoire, nous modelons l'arrivée des UPs qui suivent le loi de Poisson avec le taux d'arrivée $\alpha$. La probabilité d'avoir $k$ arrivées pendant l'intervalle d'unités de temps $t$ est

$$f(k,t) = \frac{(\alpha t)^k e^{-\alpha t}}{k!}. \tag{7.1}$$

De plus, nous supposons que les périodes de canal occupé d'UP sont exponentiellement distribuées avec le moyen $\mu$. En fait, on considère une transmission simple d'un fragment réussi, avec la probabilité $p_s$, si aucun UP n'apparaît dans ce temps ($k = 0$ et $t = L_f/R$ dans 7.1). Exprimé comme

$$p_s = f\left(k = 0, t = L_f/R\right) = e^{-\alpha L_f/R}, \tag{7.2}$$

$R$ dénote le débit binaire. Par la suite, la probabilité qu'un fragment est transmis avec succès une fois pendant les $m$ tentatives :

$$p_t = \sum_{j=0}^{m-1} p_s(1-p_s)^j. \tag{7.3}$$

Le temps de transmission de paquet réussie moyen, dénoté $\overline{T}_{tr}$, est simplement l'addition du temps de transmission réussi moyen de tous les fragments ($\overline{T}_{trF_i}$).

Comme illustré dans la figure 7.2, $\overline{T}_{trF_i}$ est composé de trois durées différentes : (i) l'intervalle de temps depuis le début de la transmission de fragment jusqu'à l'arrivée d'un UP sur le canal : $t_C$, (ii) l'intervalle de l'arrivée des UPs jusqu'à l'instant de détection suivant : $t_W$ et (iii) l'intervalle minimal depuis le début de l'instant de l'écoute que l'US exige pour rassembler assez d'échantillons pour prendre une décision correcte de la présence/absence de UP : $t_D$. La durée d'interférence $T_{int}$, dans laquelle l'US cause l'interférence à l'UP, peut être trouvée comme suit

$$T_{int} = \bar{t}_W + t_D. \tag{7.4}$$

Selon le premier terme de $\overline{T}_{trF_i}$, c'est-à-dire $t_C$, la fonction de distribution cumulative de $t_C$ peut être calculée basée sur (7.1), et la propriété sans mémoire de loi exponentielle comme suit :

$$f\left(y\right) = \frac{\alpha e^{\alpha y}}{1 - e^{\alpha L_f/R}}. \tag{7.5}$$

Par conséquent, $t_C$ est :

$$\begin{aligned} \bar{t}_C = E\left(t_C\right) &= \int_0^{L_f/R} y f(y) dy \\ &= \frac{1}{\alpha} - \frac{\left(L_f/R\right) e^{-\alpha L_f/R}}{1 - e^{-\alpha L_f/R}}. \end{aligned} \tag{7.6}$$

Nous pouvons aussi définir $t_W$ pour l'intervalle fixe $[0,\, \delta]$ selon

$$P\left(t_W \geq y \,|\, 0 \leq t_W \leq \delta\right) = \frac{P\left(y \leq t_W \leq \delta\right)}{P\left(0 \leq t_W \leq \delta\right)}, \tag{7.7}$$

$$\bar{t}_W = \delta - \frac{1}{\alpha} + \frac{\delta e^{-\alpha\delta}}{1 - e^{-\alpha\delta}}. \tag{7.8}$$

Nous notons que $t_D$ est lié au nombre minimal d'échantillons [93], $S_{min}$, que le terminal secondaire doit reprendre du signal reçu. Nous représentons la probabilité de détection correcte avec $P_d$ et la probabilité d'alerte fausse avec $P_f$, alors $t_D$ est obtenue:

$$t_D = \frac{S_{min}}{F_s} = \frac{1}{\gamma^2 F_s}\left(Q^{-1}\left(P_f\right) - Q^{-1}(P_d)\right)\sqrt{2\gamma + 1}, \tag{7.9}$$

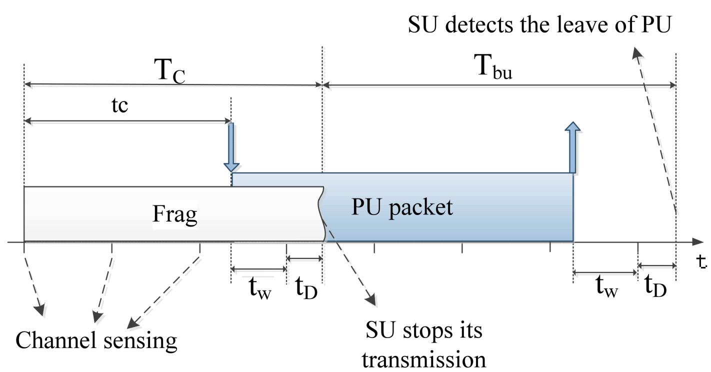



Figure 7.2: Une situation de collision.

$T_{bu}$ est simplement défini comme le temps que l'US trouve le canal occupée par l'UP. En référant à la Figure 7.2 encore une fois, il est clair que cette quantité est égale à $1/\mu$ (le temps entre l'arrivée et le départ d'UP) avec seulement un changement de $\bar{t}_W + t_D$. Donc en général, nous pouvons considérer $\overline{T}_{bu}$ comme

$$\overline{T}_{bu} = \frac{1}{\mu}. \tag{7.10}$$

Ainsi, la longueur moyenne de transmission de fragment réussie est donnée par

$$\overline{T}_{trF_i} = \sum_{j=1}^{m} \frac{p_t(1-p_t)^{j-1}}{1-(1-p_t)^m}\left(\frac{L_f}{R} + (j-1)\left(\overline{T}_C + \overline{T}_{bu}\right)\right). \tag{7.11}$$

Selon 7.11, $\overline{T}_{trF_i}$ n'est pas une fonction de $i$ , donc nous pouvons simplement l'appeler $\overline{T}_{trF}$ et, finalement, le temps total de transmission réussi de paquet serait :

$$\overline{T}_{tr} = N\overline{T}_{trF}. \tag{7.12}$$

La moyenne de fraction de données transmise réussie a été dérivée comme suit:

$$\overline{F}_T = \sum_{i=0}^{N-1} p_t^i(1-p_t)\frac{i}{N} + p_t^N = \frac{p_t - p_t^{N+1}}{N(1-p_t)}. \tag{7.13}$$

Dans la prochaine étape, la consommation d'énergie totale d'une transmission réussie de fragment $\overline{E}_{trF}$, la consommation d'énergie totale de la transmission réussie de paquet $\overline{E}_P$ et la moyenne d'énergie gaspillée à cause de baisse du paquet $\overline{E}_d$, elles sont obtenues comme suit :

$$\overline{E}_{trF} = \sum_{i=1}^{m} \frac{p_t\left(1-p_t\right)^{i-1}}{1-\left(1-p_t\right)^m}\left(E_f + (i-1)\left(\overline{E}_C + \overline{E}_{bu}\right)\right), \tag{7.14}$$

$$\overline{E}_P = p_t^N N \overline{E}_{trF}, \tag{7.15}$$

$$\overline{E_d} = \sum_{i=1}^{N-1} p_t(1-p_t)^{i-1}\left((m-1)\,\overline{E}_{bu} + m\overline{E}_C + i\overline{E}_{trF}\right). \tag{7.16}$$

$E_f$, $\overline{E}_C$, $\overline{E}_{bu}$, ils sont l'énergie consommée pendant les périodes $L_f/R$, $\overline{T}_c$ et $\overline{T}_{bu}$. Finalement, la quantité d'intérêt est le facteur de rendement énergétique de système, qui peut être calculé comme

$$\rho = \frac{E_{useful}}{E_{total}} = \frac{\overline{E}_P}{\overline{E}_d + \overline{E}_P}. \tag{7.17}$$

#### 7.1.3.2 Résultats de Simulation et Validation de Modèle

La figure 7.3 est une comparaison de la fraction des données transférées effectives, $F_T$, avant la première arrivée d'UP dans FD avec fragmentation (FFD) contre FD. Il est observé que les résultats des deux plans sont les mêmes pour des taux d'arrivée PU bas. Cependant, lorsque $\alpha$ augmente, $F_T$ diminue fortement pour le plan FD comparé au plan FFD. Cela est dû au fait que l'utilisateur secondaire FD dans un réseau occupé n'aurait pas d'occasion de transmettre ses données à tout d'un coup et le paquet entier serait baissé en présence de PU sur le canal. La fraction moyenne des données réussies qui pourraient être transmises dans un tel plan diminuerait brusquement. Quand $F_T$ diminue, le taux de chute moyenne de paquet pourrait augmenter à chaque transmission de paquet. Par consquent, cette figure ne laisse aucun doute que le plan FFD est supérieur, en particulièr pour le réseau très actif des UPs.

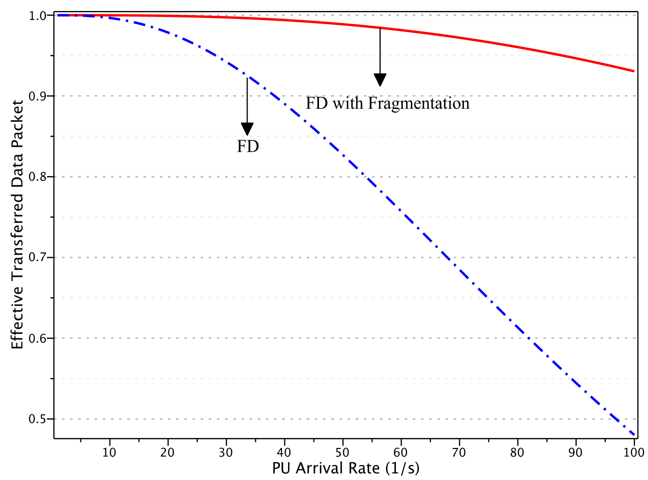



Figure 7.3: La fraction des données moyenne qui est transmise avec succès sur le canal avant la première arrivée d'PU, $\overline{F}_T$.

Finalement, le rendement énergétique des trois plans est illustré dans l'image 7.4. Ici, nous utilisons deux longueurs de données différentes pour montrer l'impact de la longueur de paquet sur performance. Clairement, il n'y a aucune grande partie de différence pour le rendement énergétique de FFD pour une longueur des données de 2000 et 7000 octets et les deux chevauchements de complots correspondant.

Cependant, les complots de FD et HD ont des changements soudains et baissent brusquement en augmentant la longueur du paquet. Pour une longueur des données de 7000 octets, nous observons que le plan HD a presque la même efficacité que le système FD et c'est aussi raisonnablement près de FFD, particulièrement pour les valeurs basses et hautes de $\alpha$. En général, il semble que le système FFD fonctionne mieux que le plan FD sans fragmentation dans un réseau à fort trafic et pour les paquets de données plus longs. Ayant presque les mêmes performances que FD et un taux de perte de paquets plus bas et en fournissant aussi moins d'interférences au

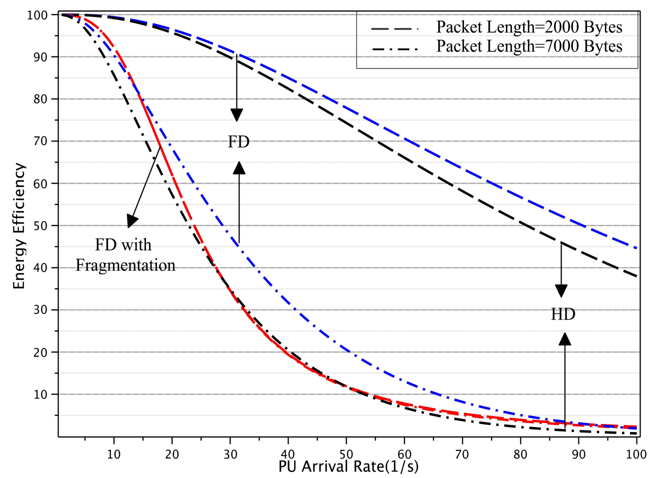



Figure 7.4: Rendement d'énergie pour les longueurs de données de 2000 et 7000 octet; HD contre. FD avec ou sans fragmentation.

PU, nous concluons que le plan FFD est un bon choix pour la génération suivante du systèmes CR.

### 7.1.4 Le Protocole DFD-MAC

Dans ce partie, en adressant aux limitations mentionnées de MAC Contra Flow [4] comme la haute probabilité de collision et la limitation sur le nombre de chemins que le paquet de données peut traverser après avoir gagné le canal, nous proposons un protocole de MAC qui démultiplie les avantages complets de la technologie full-duplex. On considère l'accès décentralisé et le protocole est appelé le MAC distribué-accès full-duplex (DFD-MAC).
Dans les réseaux décentralisés à l'étude, la technique de routage wormhole [9], [28] est utilisée en accord avec l'accès sans contention [4]. Donc, pour réduire les effets terminaux cachés, l'en-tête de données doit être envoyé avant le reste du paquet dans le trajet pour garantir la réservation du trajet avant le début de transmission des données.

De plus, le protocole proposé est capable d'expédier les paquets dans une route pour un nombre arbitraire des sauts contrairement à l'approche dans [4] qui est limité aux configurations à deux sauts seulement. Ceci est une approche efficace conçue pour prendre l'avantage complet du canal full-duplex.

Puisque le mécanisme d'accès de notre protocole de MAC proposé soit semblable au mécanisme d'accès aléatoire dans CSMA/CA. La raison de choisir CSMA comme la base de notre protocole est l'efficacité, la comestibilité et l'adaptabilité que ceci équipe le réseau avec cela. Nous modelons le DFD-MAC selon la chaîne de Markov (DTMC) similaire au DTMC de Bianchi [7].

Dans le DFD-MAC, une fois qu'un nœud sans fil termine son back-off et gagne la contention, il transmet un paquet de contrôle, appelée le paquet de demande de transfert (FRR), à sa destination d'une manière semblable que le RTS est transmis dans le mécanisme CSMA/CA. Ce paquet FRR a la même longueur ($T_{FRR}$) que le RTS et porte les mêmes informations telles que l'adresse MAC de source, l'adresse MAC de destination et le vecteur d'allocation de réseau (NAV).

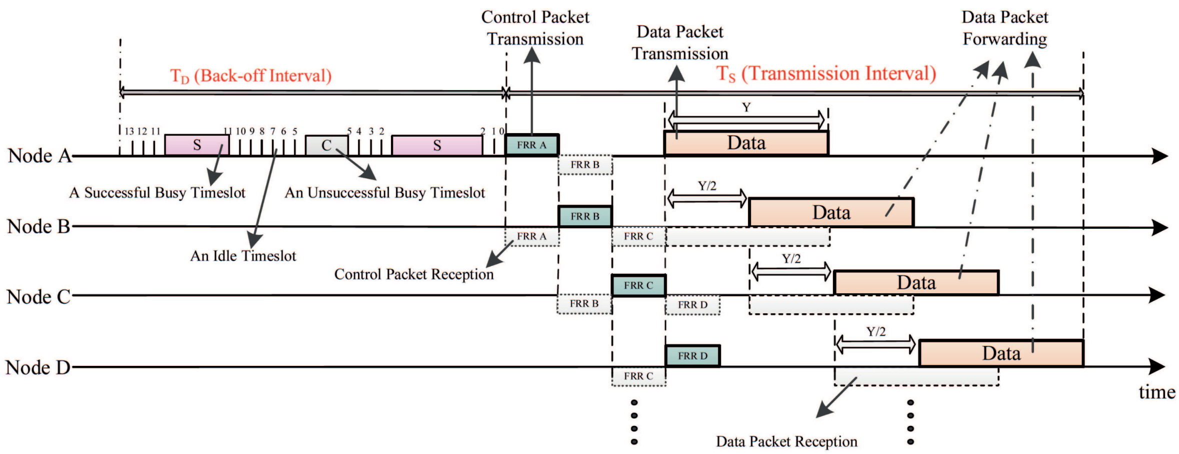



Figure 7.5: Le plan de transmission de DFD-MAC.

Nous supposons que la route est déjà déterminée par la couche réseau. Ainsi, en plus de toutes les informations que chaque FRR porte, il contient aussi l'adresse MAC du nœud suivant par lequel les données vont traverser. Finalement, les paquets FRR aident à faire rétrécir la fenêtre de collision à $T_{FRR}$ et augmenter la probabilité de transmission réussie.

La transmission dans le DFD-MAC est décomposée à deux phases :

(i) L'établissement de chemin par des paquets contrôle FRR séquentielles et (ii) la transmission des données. Comme indiqué dans l'image 7.5, pendant la première phase, après le nœud $A$ gagne la contention, il commence à transmettre le paquet FRR au nœud $B$. Une fois que le $B$ entend FRR-A, il commence à transmettre FRR-B au nœud $C$, qui compte aussi comme une acquittement implicite au nœud $A$, il l'aide à impliquer que son paquet a été déjà reçu avec succès par $B$.

La deuxième ètape commence tandis que la première phase continue encore. En fait, comme indiqué dans l'image 7.5, au cours de l'établissement du poigne de main, la transmission entre A et B commence à condition que les paquets transmis durant la premiére saut se soient éloignés de A.

Quand $B$ reçoit le paquet des données, il attend jusqu'à $T_Y/2$ secondes depuis le début de la transmission provenant de $A$ où $T_Y$ est le temps mis par le paquet des données de longueur $Y$ pour être transmis. En fait, la technologie FD autorise $B$ à transmettre en même temps que le nœud $A$ le fait, mais avec un décalage de temps d'au moins $T_Y/2$.

Notons qu'ici on n'est pas en train de traiter notre modèle de DTMC et on utilise son résultat afin de trouver la probabilité totale de transmission $\tau$. Ainsi, la probabilité de collision $p$ peut être calculée comme suit:

$$p = 1 - (1-\tau)^{N_B}(1-\tau)^{(U-1)N_{B-A\cap B}}, \tag{7.18}$$

où $U = T_{FRR}/\delta$ est le nombre de **slot** que nécessite la transmission du paquet FRR et $\delta$ est la longueur de chaque **slot**. $N_B$ $(N_A)$ est le nombre des nœuds dans la zone du transmission du nœud B (A) et $N_{B-A\cap B}$ est le nombre des nœuds dans la zone de $B-A$. En utilisant l'hypothè que la distribution des nœuds suit la loi de Poisson 2D, la probabilité que le canal soit occupé est donc égal à,

$$P_{bu} = \left(1 - (1-\tau)^{N_A}\right)\frac{N_A - 1}{N_A}. \tag{7.19}$$

En supposant qu'un nœud peut expédier seul un paquet durant la période où il a gagné la contention, la probabilité d'expédier un paquet $P_f$ est égale à:

$$P_f = \frac{\tau(1-p)}{\overline{BO}}, \tag{7.20}$$

où BO est le nombre moyen des slots de back-off inoccupés [96].

Finalement, il est nécessaire de trouver $q$ la probabilité qu'un nœud reste dans le mode d'expédition. Nous n'allons pas montrer les détails de cacul de $q$ dans cette partie. Les lecteurs pourraient se référer au chapitre 4 pour plus de détails. Nous obtenons donc

$$q = \frac{\overline{T}_{s,Fw} - \delta}{\overline{T}_{s,Fw}}, \tag{7.21}$$

où $T_{s,Fw}$ est

$$T_{s,Fw_i} = (3-n)\,T_{FRR} + \frac{n+2}{2}Y, \tag{7.22}$$

avec $n$ comme le nombre moyen de sauts que le paquet de données peut parcourir.

#### 7.1.4.1 Analyse de Performance du DFD-MAC

On définit le débit de trajet normalisée $TH_{path}$ comme les données utiles qui peuvent être transmises pendant une période de temps (dénotée $T_t$ pour le moment) :

$$TH_{path} = \frac{E(Y)}{T_t}. \tag{7.23}$$

Pendant l'intervalle de temps, appelé le retard de trajet ($D_{Path}$), une quantité fixe d'informations est certainement livrée ($E(Y) = cte$). Ceci peut être considére comme une définition alternative pour la débit du transmission. Par conséquent $T_t = D_{Path}$, où

$$D_{Path} = T_D + T_S. \tag{7.24}$$

Comme indiqué dans l'image 7.5, basée sur l'approche dans [96], le temps de la contention, $T_D$, est composé de quatre intervalles. Ces derniers sont : le temps de collision gaspillé de A, $T_C$, pendant lequel les essais de transmission de A ont échoués ; (ii) le temps de transmission réussie total des voisins $T_{S'}$, pendant lequel nœud A se met en pause puisque le canal est occupé avec les transmissions réussies des autres ; (iii) le temps de collision total des voisins $T_{C'}$, pendant lequel le nœud A se met en pause puisque le canal est occupé avec des transmission infructueuses des autres ; et (iv) le temps de repos $T_I$, pendant lequel le canal n'est pas occupé. Ainsi, $T_D = T_C + T_{S'} + T_{C'} + T_I$.

Après le calcul de tous les temps inconnus nous sommes en mesure d'évaluer le débit et le retard du plan DFD-MAC.

#### 7.1.4.2 Validation du Modèle

Dans l'image 7.6 et l'image 7.7, nous comparons le retard du trajet et les résultats de débit du trajet en considérant les nombres différents des nœuds voisins (montré comme $N_N$ dans les figures).

Selon l'image 7.6, quand le nombre de nœuds voisins diminue, le retard du trajet du protocole CSMA/CA et de MAC FD diminuent. Ceci est prévisible puisque la probabilité de collision augmente toujours pour un réseau plus dense. Le paquet fait plus de retard lors de son acheminement. Cependant, nous pouvons constater que la performance de notre protocole MAC est meilleure que CSMA/CA dans chaque cas et qu'il devient encore plus performant que ce dernier quand le nombre moyen de sauts augmente. Ceci s'explique par les chevauchements entre les transmissions des différents sauts et la chance d'avoir la transmission sans contention pour les nœuds d'expédition. Le même raisonnement peut être appliqué pour l'image 7.7 pour justifier l'amélioration de performance du DFD-MAC qui est deux fois mieux en moyenne.

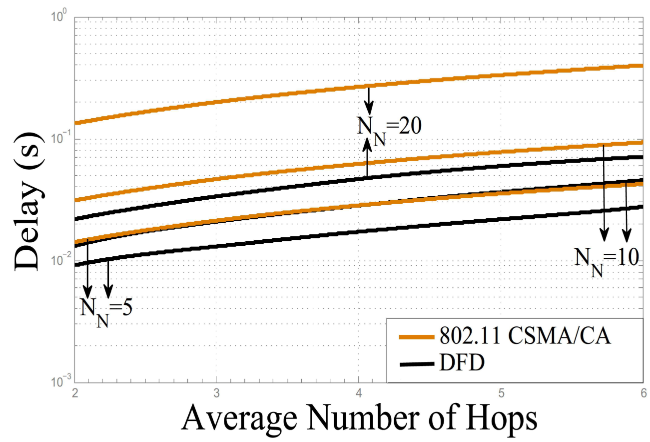



Figure 7.6: Le retard du chemin dans DFD-MAC et 802.11 CSMA/CA pour des nombres différents de noeuds voisins.

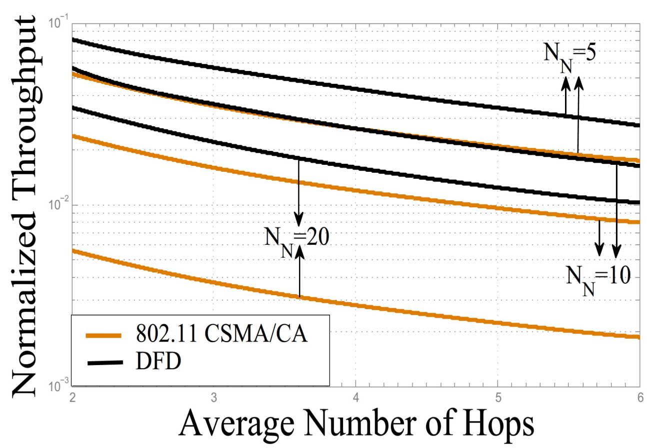



Figure 7.7: Le débit du chemin dans DFD-MAC et 802.11 CSMA/CA pour des nombres différents de noeuds voisins.

Finalement, en comparant les deux protocoles, nous pouvons déclarer que la performance de notre protocole est beaucoup meilleure que celle de 802.11 bien que sa collision de paquet et sa probabilité d'occupations de canal est plus haute. Ceci est grâce à la technologie FD qui permet de supporter le chevauchement des transmission durant les sauts consécutifs.

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